Searches for User *jcorrielus1* (Count = 12331)

Queries 12282 through 12331.

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S #	Updt	Database	Query	Time	Comment
S12331	U	USPT	(lightweight near directory near access near protocol) and subscriber and telephone and (subscriber near director$) and update	2005-05-20 14:22:01	
S12330	U	USPT	(lightweight near directory near access near protocol) and subscriber and telephone and (subscriber near director$)	2005-05-20 14:21:36	
S12329	U	USPT	(lightweight near directory near access near protocol) and subscriber and telephone	2005-05-20 14:21:17	
S12328	U	USPT	(lightweight near directory near access near protocol) and subscriber	2005-05-20 14:21:08	
S12327	U	USPT	(lightweight near directory near access near protocol)	2005-05-20 14:20:54	
S12326	U	USPT	(lighweigth near directory near access near protocol)	2005-05-20 14:20:21	
S12325	U	USPT	6498791.pn. and (lighweigth)	2005-05-20 14:19:47	
S12324	U	USPT	6498791.pn. and ldap	2005-05-20 14:19:19	
S12323	U	USPT	6498791.pn.	2005-05-20 14:18:25	
S12322	U	USPT	updat$ near subscriber near director$ and (automatic$) and telephone and address and (area near code)	2005-05-20 10:11:33	
S12321	U	USPT	updat$ near subscriber near director$ and (automatic$) and telephone and address	2005-05-20 10:11:09	

Searches for User *jcorrielus1* (Count = 12322)

Queries 12273 through 12322.

Find

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S #	Updt	Database	Query	Time	Comment
S12322	U	USPT	updat$ near subscriber near director$ and (automatic$) and telephone and address and (area near code)	2005-05-20 10:11:33	
S12321	U	USPT	updat$ near subscriber near director$ and (automatic$) and telephone and address	2005-05-20 10:11:09	
S12320	U	USPT	updat$ near subscriber near director$ and (automatic$) and telephone	2005-05-20 10:10:47	
S12319	U	USPT	updat$ near subscriber near director$ and (automatic$)	2005-05-20 10:10:27	
S12318	U	USPT	updat$ near subscriber near director$ and (subscriber near message)	2005-05-20 10:09:52	
S12317	U	USPT	updat$ near subscriber near director$	2005-05-20 10:09:20	
S12316	U	USPT	autonomous near telephony and (telephone near number) and controlling and updat$	2005-05-19 21:48:41	
S12315	U	USPT	autonomous near telephony and (telephone near number) and controlling	2005-05-19 21:48:19	
S12314	U	USPT	autonomous near telephony and (telephone near number) and controlling3	2005-05-19 21:48:14	
S12313	U	USPT	autonomous near telephony and (telephone near number) and routing	2005-05-19 21:47:50	
S12312	U	USPT	autonomous near telephony and (telephone near number)	2005-05-19 21:47:17	
S12311	U	USPT	autonomous near telephony	2005-05-19 21:45:31	

L9: Entry 2 of 5 File: USPT Apr 19, 2005

DOCUMENT-IDENTIFIER: US 6882708 B1
TITLE: Region-wide messaging system and methods including validation of transactions

Abstract Text (1):
Methods and systems are disclosed that allow for the exchange of voice mail messages between different VMSs of different service providers and/or in different networks by the transmission of such messages through a data network using a standard protocol of the data network. Methods and systems also are disclosed that validate message transactions among subscribers receiving regional messaging services over the PSTN. The subscribers are located in different geographic areas and may be provided their voice, facsimile or data messaging services by different companies. The present invention validates passing messages (data) among customers of potentially different companies located in different areas by assessing the validity of the transaction in light of a number of conditions, including applicable regulatory or business conditions.

Brief Summary Text (4):
Telephone answering machines are used by many consumers to collect messages that are received while the consumers are unavailable. But such answering machines have limitations that pose inconveniences. For example, a conventional telephone answering machine generally will not take a message from a caller when the called party is already engaged in a call. The caller must call again even though the called party has an answering machine. Some of the limitations of telephone answering machines have been overcome by network voice mail services typically offered by telecommunications service providers. For example, generally, a network voice mail service will take a message from a caller when the called party is already engaged in a call.

Brief Summary Text (5):
While telephone answering machines and network voice mail services are used by consumers in the home and in small businesses, other telecommunication products have been developed to serve larger businesses, and other institutions such as schools, hospitals, government offices, and the like. These other telecommunication products include telecommunications systems having advanced messaging features. These advanced messaging features typically provide a user with more options than a conventional telephone answering machine or network voice mail service.

Brief Summary Text (9):
Considering the three types of products discussed (telephone answering machines, network voice mail services, and telecommunications systems having advanced features), there are needs of users left unsatisfied by these products. For subscribers to network voice mail services, and especially for users of telephone answering machines, there is a need for an apparatus, system, or method that will provide the functionality of the telephone answering machines and the network voice mail services as well as provide the advanced messaging features of the telecommunications systems generally used in larger institutions. But it is not enough to satisfy the needs of users by providing telecommunications systems having advanced features to subscribers of network voice mail services and/or users of telephone answering machines. It is not satisfactory because the advanced messaging features of such systems are available only to persons associated with the institution having deployed the particular telecommunications system. Thus, there is a need for an apparatus, system, and/or method that provides a user with advanced messaging features and that may be used in connection with communications to other users even if the other users are not associated with a common institution. In sum, there is a need for an apparatus, system, and/or method that implements a messaging system across a region for the exchange of communications between and among users of the region.

Brief Summary Text (10):
Multiple obstacles exist to providing users with a region-wide and feature-rich messaging

system. The region-wide messaging system may include multiple service providers with each service provider having one or more voice mail platforms, etc. As a result, technical, regulatory, and business constraints may prevent the exchange of messages in the region-wide messaging system between users who reside in different states or areas of the region, and/or who are served by different service providers. For instance, in the United States, some state and/or federal regulations prevent certain categories of service providers from transferring telephone calls, and possibly messages, across state boundaries and/or across LATAs ("Local Access Transport Areas"). Also, even if users are in the same state or LATA, a user may choose to subscribe to messaging service from a service provider different from another user's service provider. Unless service providers have reciprocal business agreements for accepting each others' traffic within the region-wide messaging system, message exchange between the users may not be possible or may be possible only accompanied by a large toll charge. Similar regulatory restrictions and business considerations may apply in other countries and/or in messaging systems that operate across national borders and are included in a region-wide messaging system.

Brief Summary Text (11):
Some of the obstacles posed to a region-wide messaging system by regulatory, technical, and business constraints can best be illustrated by an example. Assume two people (Oscar and Rachel) subscribe to voice mail service provided through a region-wide messaging system. Oscar lives in Louisiana. In Louisiana, BellSouth provides Oscar with local telephone service, and Oscar has chosen AT&T to provide voice mail service through the region-wide messaging system. Oscar wishes to originate or leave a message for Rachel, the message's recipient who lives in Georgia. In Georgia, BellSouth provides Rachel with local telephone service, and Rachel has chosen BellSouth to provide voice mail service. Oscar calls Rachel, who is unavailable, so Rachel's voice mail service plays a pre-recorded message and prompts Oscar for his message. Oscar leaves a message. Rachel eventually retrieves the message, but instead of actually talking with Oscar, Rachel would prefer to dash off a quick reply by using her voice mail service.

Brief Summary Text (20):
This invention a region-wide messaging system that uses more than one messaging server ("MS"), such as a voice mail platform. Each MS may be located respectively in a different geographic region and operated by a different service provider. A caller initiates a message in a first MS to be delivered to a subscriber served by a second MS. The first MS queries a directory by forwarding at least the subscriber's telephone number. Using the telephone number of the recipient (subscriber), as well as either the sender's phone number or information identifying the first MS that the first MS forwards to the directory, the directory applies various tables and rules to determine the service provider and location (by, e.g., LATA, state or other subdivision) for each of the sender and recipient. If regulations allow messaging transactions between those locations, and if the first and second MSs are operated by different service providers, then a determination is made as to whether one or more agreement(s) exist(s) between the different service providers that will allow the transaction to be validated and go forward. These determinations can, of course, also be carried out after the messaging transaction in order, for instance, to determine the billing rate for the particular messaging transaction.

Brief Summary Text (21):
A region-wide messaging system may also allow subscribers to activate a message delivery service for the delivery of messages to a group of recipients, including subscribers and non-subscribers of service providers associated with the region-wide messaging system. A message to a subscriber is delivered only if the proposed transaction associated with the message is validated. A message to a non-subscriber need not be validated because such a message is delivered by a telephone call to the non-subscriber's voice mailbox, a function allowed under existing regulations.

Brief Summary Text (22):
Whether replying, forwarding or distributing a message to one or more subscribers or non-subscribers, the destination address (e.g., telephone number) is collected. For a reply message to a non-subscriber, the non-subscriber's address (e.g., telephone number) may be discerned by analyzing the calling line identification information provided by the telephone network when the non-subscriber leaves the original message. For a reply message to a subscriber, the subscriber's telephone number is collected from the originating address (e.g., telephone number) of the message left by the subscriber.

Brief Summary Text (23):
As noted, the destination telephone number as well as either the originating telephone number of MS identity are provided to the directory, which allows the present invention thereafter to determine whether business and regulatory rules allow the messaging transaction to go forward. Additionally, the destination telephone number allows a verification message to be provided to the user originating the transaction. The verification message alerts the user both to the validity of the transaction and to the identity of the message's recipient. For subscribers, the verification message may be formulated by using the destination telephone number to retrieve, via the directory, the subscriber's spoken name or other audio message; for non-subscribers, the verification message may simply be the telephone number.

Brief Summary Text (24):
One type of region-wide messaging system with which the present invention may be used is deployed over a public switched telephone network (PSTN) that uses various Advanced Intelligent Network ("AIN") components. For instance, commercially available voice mail platforms may be reconfigured as messaging servers and provided with appropriate AIN functionality so that the messaging servers act as an intelligent peripheral within the AIN. The messaging servers interface with another, directory server, such as Lightweight Directory Access Protocol ("LDAP") directory servers available from various suppliers and which may be configured to hold the directory information. Or, Service Control Points ("SCPs") could be reconfigured with LDAP server functionality and adapted to hold the one or more directories and databases that are used with the present invention. In any event, each directory server may be equipped with a Regional Messaging Directory ("RMD") that indexes telephone numbers to identify the messaging server (MS) serving a particular telephone number or group of telephone numbers. One or more other tables in the directory may list: (a) the location of particular MSs, by LATA, state or other subdivision; (b) the rules governing message transactions among inter-state, inter-LATA or other inter-divisional MSs; (c) the business agreement rules governing the exchange of messages between service providers and others associated with the region-wide messaging system; and (d) certain flags that indicate whether messages may be sent to or from a particular subscriber based on a variety of criteria ranging from whether the subscriber has paid his bills to a subscriber's particular language.

Brief Summary Text (25):
A preferred embodiment of a region-wide messaging system in which the present invention may be deployed uses a transmission control protocol/internet program (TCP/IP) network to allow transfer of messages among various MSs. Queries to and responses from the LDAP Server on which the RMD resides may be executed using internet protocols, such as the Lightweight Directory Access Protocol ("LDAP") that is a TCP/IP-based derivative of the X.500 electronic mail delivery service. Skilled persons will recognize that while a region-wide messaging system may use TCP/IP and LDAP protocols to query directories and other network elements, other protocols may be used, such as the signaling system 7 (SS7). Indeed, SS7 protocol would allow implementers to take advantage of the reliability and known capability of an established telecommunications protocol.

Brief Summary Text (26):
In another aspect of the invention, a region-wide messaging system is facilitated by allowing subscribers to reply to received messages by authenticating the reply recipient and the proposed reply transaction. The subscriber's MS queries the LDAP Server's RMD via messages in LDAP protocol that provide the RMD both the subscriber's telephone number and the reply recipient's telephone number. (As used herein, the telephone number is assumed to identify the voice mailbox, although other identifiers may be used). The RMD determines the identity of each MS associated with the telephone numbers provided. Using a number of tables, the RMD ascertains the geographic location of each MS and whether regulatory rules allow a message transfer between MSs in those locations. Assuming a positive response, the RMD determines whether, if the MSs are operated by different service providers, those service providers' agreements allow exchange of messaging traffic. The RMD provides the subscriber's MS with this information, and the MS, in turn, authorizes or stops the proposed messaging transaction.

Brief Summary Text (27):
Optionally, the present invention may determine whether a reply message is possible before the subscriber attempts such a reply. For example, when a subscriber accesses the voice mail service for messages, a message is played to the subscriber. During message retrieval, the

subscriber's MS queries the RMD to determine whether a reply may be made to the message originator, using generally the same procedure as described above. After validation of a reply, the MS informs the subscriber that a reply may be made, perhaps through an announcement of "Would you like to reply to [spoken name of originator]?"

Brief Summary Text (28):
Similarly, when a subscriber wishes to formulate a message for multiple recipients, the subscriber accesses the voice mail service, prepares the message and enters the list of destination telephone numbers. The subscriber's MS formulates an LDAP query that provides the LDAP Server with the telephone number(s) of the recipient(s) for the message, as well as the subscriber's telephone number. Upon receiving those numbers, the LDAP Server determines whether they are numbers of subscribers who have voice mail service through the region-wide message system. The LDAP Server then routes the numbers to the RMD. The RMD compares the numbers to an index and determines the identity of the serving MS for each number. Using other tables and indexes, the RMD determines the location of each MS, whether regulatory rules permit messaging transfers between the subscriber's (i.e., the message sender) MS and the recipient's MS, and whether, if the two MSs are operated by different service providers, business agreements between the service providers allow the recipient's MS to accept messages from the subscriber. That information is returned to the subscriber's MS, which alerts the subscriber to any telephone numbers to which messages may not be delivered. For validated telephone numbers, the messaging transaction proceeds.

Brief Summary Text (29):
Validation of messaging transactions may be adapted to particular situations. For instance, if the region-wide messaging system is deployed in multiple countries, only some of which allow international messaging transactions, the present invention can be adapted to validate those transactions also. Further, it is anticipated that the validation criteria may be modified or changed as regulations from regulatory bodies and agreements among service providers change. Indeed, the examples above are only a few of the ways in which proposed message transactions may be validated. Other validation criteria may be selected. For instance, the directory may indicate a subscriber has paid his or her bills and can proceed with messaging transactions (e.g., if the subscriber has not paid the system may reject the message. Another important billing feature may be to alert the user to whether the message will incur an additional fee, for instance because the message exceeds the number of messages purchased by the user for that particular billing cycle. Or, the directory may indicate the sender's and receiver's spoken language or other shared features, which the present can compare. Another important criteria on which to validate the message transaction may be to determine whether the user has paid his bill or is otherwise authorized to use the messaging service. These and other validation criteria may be added to the directory. Thereafter, the present invention validates the transaction based on the new or added validation criteria.

Brief Summary Text (30):
Although the embodiment described above and elsewhere in this document describes deploying the directory upon an LDAP Server, persons skilled in this field will recognize that other platforms can be used to perform that functionality. For instance, the directory can be integrated into a workstation that in turn can couple to the public switch telephone network ("PSTN"). Additionally, the present invention can be adapted for use with IP addresses instead of telephone numbers.

Brief Summary Text (32):
To provide a method or system for providing messaging subscribers a convenient and reliable way of exchanging information with other users of the region, whether or not the other users are subscribers associated with the region-wide messaging system.

Brief Summary Text (39):
To provide a method or system capable of either blocking messaging services for or to certain subscribers or billing certain subscribers for particular messaging transactions.

Detailed Description Text (5):
Caller: A caller is a participant in a messaging transaction who has placed a telephone call that may result in a message to a subscriber of a region-wide messaging system. A caller may also subscribe to the region-wide messaging system, offered either through the same service provider as that of the subscriber or through another service provider.

Detailed Description Text (9):
Subscriber: A subscriber is a person or entity who receives the benefit of services offered by service providers participating in a region-wide messaging system. The subscriber need not necessarily be the person who actually pays for the services.

Detailed Description Text (10):
Transaction: A transaction is the transfer of a message from one originating device or messaging server (MS) to a destination device such as a telephone or computer or another MS, which may be a different or same type of MS, located in a different or the same region, or operated by a different or the same service provider. The message may be a reply message, a message formulated to go to one or more recipients, a forwarded message (whether fax, another voice mail, video or data), or any other type of message transmitted among the MSs of the region-wide messaging system and intended for review by a desired recipient.

Detailed Description Text (12):
Messaging Server: A messaging server (MS) is a platform, including both hardware and software, from which voice mail and other messages and other services involving message transfer, reply, forwarding, etc. are provided to subscribers. The inventions described herein are not restricted to a particular embodiment of voice mail or other messaging server since it is fully intended that different types of voice mail or messaging servers, perhaps operated by respectively different service providers, may be used within and without a region-wide messaging system for messaging transactions.

Detailed Description Text (15):
The RWM system described herein may allow a subscriber to the messaging system within the region of the service provider to send, receive, forward, and reply to messages, including voice mail messages and Voice Profile for Internet Mail (VPIM) Messages. Subscribers may receive messages from other subscribers and non-subscribers. Subscriber-to-subscriber messaging, however, illustrates the advanced features of the RWM system, which may be available, such as: (1) each subscriber may send a message to another subscriber; (2) each subscriber may reply to a message received from another subscriber; (3) each subscriber may reply to a telephone message received from a non-subscriber by implementing a feature that dials the non-subscriber; and (4) each subscriber may receive and reply to internet voice messages or fax messages.

Detailed Description Text (17):
FIG. 1 is a block diagram of an exemplary RWM system 10 (also referred to as a telecommunications messaging network). The network 10 includes a variety of interconnected network elements. A group of such elements includes the plurality of end offices which are indicated as service switching points (SSPs or switches) 12a, 12b, 12c. An SSP typically includes switch functionality, but also includes other functionality so as to communicate with other network elements, and in particular, with Advanced Intelligent Network (AIN) elements. SSP 12a and SSP 12c are each coupled to a subscriber line, which also may be referred to as a line or a calling line. Each SSP 12a, 12b, 12c serves a designated group of lines, and thus, the SSP that serves a particular line may be referred to as its serving switch. The line is typically connected to a piece of terminating equipment including telephones 14, 38. Although telephones 14, 38 are illustrated as the terminating equipment, those skilled in the art will understand that such terminating equipment may include other telecommunications devices including, but not limited to, facsimile machines, computers, modems, etc. End offices may further be coupled through a tandem office (not illustrated), which may be used to connect and switch circuits between and among end offices.

Detailed Description Text (18):
Each active line in an AIN is assigned a ten digit (NPA-NXX-XXXX) line number regardless of whether seven or ten digits are dialed to reach the subscriber. A line number is commonly referred to as a telephone number or a directory number.

Detailed Description Text (20):
SSPs 12a, 12b, 12c are interconnected by a plurality of trunk circuits 18. These are the voice path trunks that connect the SSPs to connect communications. In addition to connections to other elements, each of the SSPs is connected to a local signal transfer point (STP) 20 via respective data links. Currently, these data links employ a signaling protocol referred to as

Signaling System 7 (SS7). Much of the intelligence of the AIN resides in a service control point (SCP) 22 that is connected to STP 20 over an SS7 data link. Among the functions performed by the SCP 22 is the maintenance of network databases and subscriber databases as represented collectively by databases (subscriber data) 24.

Detailed Description Text (21):
In order to keep the processing of data and calls as simple as possible, a relatively small set of triggers is defined at the SSPs for each call. A trigger in the AIN is an event associated with a particular call that generates a packet to be sent to an SCP. The SCP queries its databases or service package applications (SPAs) for processing instructions with respect to the particular call. The results are sent back to the SSP in a response from the SCP 22 through STP 20. The return packet includes instructions to the SSP as to how to process the call. The instructions may be to take some special action as a result of a customized calling service or an enhanced feature. In response to the instructions, the SSP moves through the remaining call states, may encounter further triggers, and generates further packets that are used to set up and route the call. Similar devices for routing calls among various local exchange carriers are provided by regional STP (not illustrated) and by regional SCP (not illustrated) which may be connected to STP 20, SCP 22, and/or to the elements described herein through the public switched telephone network (PSTN) 26.

Detailed Description Text (22):
When a messaging subscriber (such as the person or entity using telephone 14) subscribes to a messaging service, an entry or a record is created in a VMS such as VMS 15. Each VMS 15, 17 includes subscriber administration, message retrieval, send, reply, forward, and mailbox maintenance functions, among others. Each VMS 15, 17 includes or is functionally connected respectively to a subscriber profile database 28, 30 (subscriber data). Each subscriber profile database stores subscriber-specific profile information (subscriber information) for retrieval by VMS functions. The VMSs 15, 17 are elements of the messaging system or service. To the protected TCP/IP network(s) 32 described below, each of the messaging platforms 15, 17 look like a valid TCP/IP element. In support of this, the VMSs 15, 17 may be assigned a TCP/IP (or IP) address and/or a domain name. Generally, the TCP/IP or other address or domain name of the VMS 15, 17 may be stored in a region-wide messaging directory (RMD) 25 discussed below, or may be stored on some domain name server (not illustrated) either in the protected TCP/IP network (s) 32, in some other element, or as a separate element. In further support of this TCP/IP capability, the VMSs 15, 17 may also provide operations access to mail administrative destinations, in addition to subscriber messaging mailbox destinations. In addition, each VMS 15 or 17 is an SS7 network element and as such is assigned an identifier such as a directory number, a destination point code (DPC) or the like.

Detailed Description Text (23):
The VMSs 15, 17 communicate with the SSP and the SCP according to the AIN 0.2 Switch--Intelligent Peripheral Interface Generic Requirements--1129-CORE Specification, AINGR: Switch--Intelligent Peripheral Interface (IPI) (A module of AINGR, FR-15); Document Number: GR-1129; Issue Number: 03; Updates: REV01--October 1998; Issue Date: September 1997; Product Type: Industry Requirements and Standards (RS); Component of FR-15, ("GR-1129") which is incorporated herein by reference. This GR-1129 describes the use of a Remote Operations (RO) parameter for indicating the invocation of a supplementary service. The RO parameter may be used to allow the SCP 22 and the VMSs 15, 17 to share information. If the caller or the communication desires to exercise an option of an action other than leaving a message, such as an attempt to contact the subscriber at the different directory number, the caller or communication provides the indication of the action to be taken with respect to the communication. For example, the caller may press "0". In response to receipt of the indication of the action by the VMS 206, a transmitter (not illustrated) of the VMS 206 transmits a message indicating the action to be taken with respect to the communication and indicating a release of the communication by the VMS 206. The message may be an GRU-1129 message including a remote operations (RO) parameter. The RO parameter may include information indicating what action is to be taken with respect to the communication such as a transfer of the communication (away from the VMS 206). This information may be stored in a field of the RO parameter such as a field denominated as an "operation type" field.

Detailed Description Text (26):
Advantageously, a subscriber's line number generally may be the subscriber's mailbox number associated with a messaging platform rendering service to the subscriber in the RWM system. In

other words, a message addressed to the subscriber may include the subscriber's line number, which may also be the subscriber's mailbox number. Alternatively, the subscriber's mailbox number may relate to some other identifier associated with the subscriber. The subscriber's address may be based on the ten digit directory number (DN) using an International Telecommunications Union (ITV) Standard E.164 compliant address.

Detailed Description Text (27):
FIG. 1 also illustrates the exemplary use of a region-wide messaging directory 25 (RMD or directory) in the messaging system 10. The RMD 25 is functionally connected to the other elements of the messaging system 10 through inclusion in or a connection to the TCP/IP network 32. Although the RMD 25 is illustrated as connected to the system 10 through the TCP/IP network 32, the RMD 25, or course, may be connected to the system 10 in other ways or even be included in an element of the system such as in association with the directories 24 of SCP 22. An RMD provides high-speed directory look-up for messaging subscribers. Generally, an RMD stores information so as to determine which messaging platform of the RWM system serves which subscriber. Additional information on the manner in which the RMDs of the messaging system 10 store information on messaging platforms and subscribers and how RMDs interact with a network element 51 may be obtained from the commonly assigned and owned patent application entitled "Methods and System for Determining Message Routing Based on Elements of a Directory Number", which was filed with the United States Patent and Trademark Office on Dec. 13, 1999 and assigned Ser. No. 09/459,498, and which was filed with the United States Receiving Office pursuant to the Patent Cooperation Treaty (PCT) on Dec. 13, 1999 and assigned Application No. PCT/US99/29491 and which application is herein by reference.

Detailed Description Text (28):
Of course, an RMD may keep track of other information relating to subscribers of the RWM system. In particular, the RMD may act as both a client and a server with respect to the Light-weight Directory Access Protocol (LDAP). The RMD stores subscriber, service, and other messaging data. In addition, the RMD supports the LDAP attributes field for LDAP clients to choose the fields that they desire to retrieve from the server. Clients may retrieve the subscriber profile from the RMD.

Detailed Description Text (29):
Subscriber data may be stored in the RMD in the following exemplary fashion:

Detailed Description Text (30):
Subscriber data is used to look up subscribers in the RMD. The data is also used for the purposes of routing and billing a subscriber's calls and messages to and from the messaging platforms.

Detailed Description Text (32):
The service data contains messaging platform-specific information to perform certain checks during directory look-up and call/message routing. The RMD may also store service provider data to ensure that a service provider has access to only its authorized subscribers' information.

Detailed Description Text (36):
The RWM system 40 of FIG. 2 includes a data network 42, which may be the internet, an intranet, or other data network using at least one standard protocol to transmit messages through the data network. A standard protocol may be the Transmission Control Protocol/Internet Protocol (TCP/IP), the Lightweight Directory Access Protocol (LDAP), which is a TCP/IP-based derivative of the X.500 electronic mail (e-mail) delivery service, Profile for Internet Mail (VPIM) protocol, or the like.

Detailed Description Text (37):
In the exemplary RWM system 40, the data network 42 is connected to a network A 44 such as a segment of the public switched telephone network (PSTN) or similar network. Network A 44 includes a voice mail server (VMS) A 46, which is operated by a service provider to provide messaging services to subscribers such as user A 48.

Detailed Description Text (38):
The data network 42 also is connected to a network B 50, which may be a different segment of the PSTN or similar network from that segment of the PSTN represented by network A 44. Network B 50 includes two voice mail servers (VMSs) 52, 54. In this example, each VMS 52, 54 is

operated by a different service provider from the other and from the service provider of VMS 46 in network A 44. Each VMS 52, 54 is operated by its service provider to provide messaging services to a respective group of subscribers. VMS B 52 provides messaging services to user B 58. Network B 50 also includes a directory 56 such as a Regional Messaging Directory (RMD) described above or similar directory. Network B 50 uses directory 56 to determine the address of or other routing information for a message received from the data network 42. In this example, the directory keeps track of which users (telephone numbers, directory numbers, addresses, or the like identifiers) are served by which of the two VMSs 52, 54.

Detailed Description Text (44):
FIG. 3 shows that SCPs 226, 228 are each coupled to a regional messaging directory ("RMD") 240, 242, respectively. RMDs 240, 242 may also act as an LDAP server responding to LDAP clients (like VMSs), able to respond to various LDAP queries by replying with the information indicated in the LDAP fields. Generally, RMDs 240, 242 provide a central location for subscriber information management. Initially, the RMDs 240, 242 may store only subscribers' 212 e-mail and SMTP server addresses, but they may contain placeholder attributes or pointers for information presently stored in VMSs 244, 246 such as a subscriber's name announcement (a.k.a. Spoken Name), extended absence greeting indicator and sub-mailboxes. RMDs 240, 242 may be physically deployed on an SCP 226, 228 or, preferably, may reside on a server (computer) and be linked to the SCPs 226, 228 via an appropriate network connection. RMDs 240, 242 may be configured as an Oracle.TM. database available from the Oracle Corporation deployed on an AIN SCP available from Lucent. Other reliable platforms and databases may be used to implement RMDs, 240, 242, including UNIX-based or WindowsNT servers.

Detailed Description Text (45):
As noted above, each RMD 240, 242 may store various types of information. For example, each RMD 240, 242 stores and maintains subscriber profile information that may consist of the types of information set forth in the exemplary tables above. SCPs 226, 228 access RMDs 240, 242 in order to respond to LDAP messaging queries by providing the information requested by the inquiring VMS 244, 246. This information can include: validation of the number provided by the subscriber, other subscriber information like spoken name, etc. or the addresses of the network element from which such information may be retrieved. Subscriber data is used to look up subscribers in the RMD. The data is also used for the purposes of routing and billing subscriber calls to and from the messaging platforms.

Detailed Description Text (46):
Service data also may be stored in the RMD as illustrated by the tables above. The service data contains messaging platform-specific information to perform certain checks during directory look-up and call routing. The RMD may also store service provider data to ensure that a service provider has access to only its authorized subscribers' information.

Detailed Description Text (48):
A Subscriber Table stores the information concerning the State and LATA (or other appropriate political, regulatory or geographical area) in which each VMS resides, as well as the service provider operating that VMS.

Detailed Description Text (52):
VMSs 244, 246 may use LDAP for retrieving subscriber profile information. Both RMDs 240, 242 and VMSs 244, 246 may interchangeably act as a LDAP client as well as a directory server. RMDs will support primary and secondary LDAP queries so that initially destination information can come from RMDs 240, 242 (via a primary query) and spoken name and other subscriber information can come from other VMSs (via secondary queries) within the system 200. VMSs 244, 246 may query RMDs 240, 242 for subscriber profile information, essentially acting as a LDAP client. VMSs 244, 246 will request all subscriber attributes, but RMDs 240, 242 will return only usable values for email and SMTP addresses. When VMSs 244, 246 receive the attributes that are unusable values but represent placeholders for such profile information as the spoken name, sub-mailboxes etc., an originating VMS 244, 246 will act as a server, initiating a secondary query to a particular, client VMS for that information. For instance, VMS 244 may act as a LDAP client, retrieving subscriber information like the subscriber's spoken name announcement by querying RMD 240, which either returns the spoken name or returns information indicating VMS 246 retains the subscriber's spoken name. VMS 244 thereafter sends a secondary query via VMS network 252 to VMS 246 to retrieve the stored spoken name.

Detailed Description Text (53):
FIG. 4 shows the general network topology and interrelation between a VMS 246 acting as an LDAP client seeking subscriber information from RMD 240, which in turn queries a VMS 244 acting as an LDAP server to retrieve a spoken name. Alternatively, all subscriber information may be stored in and retrieved from RMDs 240, 242, which may also retrieve subscribers' spoken names directly from VMSs. VMS 244, 246 queries to RMDs 240, 242, may use the domain name scheme described in the AIM2000 Service Deployment Architecture, attached and previously incorporated by reference. Additionally, detailed call flows showing the various LDAP queries and responses among the network elements shown in FIGS. 3 and 4 are described in further detail in the "AIM2000 Service Deployment Architecture (Issue 1.0)" document, attached to the referenced provisional application and previously incorporated by reference.

Detailed Description Text (54):
Using the LDAP query scheme described above, the present invention facilitates VMSs 244, 246 access to subscriber profile information stored in RMDs 240, 242. For instance, when a message is to be delivered among subscribers to system 200, one VMS 244 will query the appropriate RMD, providing the destination and origination addresses (e.g., telephone numbers). The RMD 240 will associate those addresses with the applicable VMS serving those subscribers, which will also provide information concerning the geographic location of the VMS and the service provider operating the VMS. Then, using rules provided within its database, RMD 240 will determine whether regulatory, business or other constraints prohibit a messaging transaction between the identified VMSs of the message sender and recipient.

Detailed Description Text (56):
Certain subscribers may elect to use a message delivery service that allows subscribers 200 to create a multiple distribution list by which messages can be broadcast to multiple recipients. The distribution list can include other subscribers to regional messaging system 200 and non-subscribers (although the non-subscribers must have at least the traditional voice mail services).

Detailed Description Text (57):
As the subscriber 212 creates the distribution list by inputting various telephone numbers, validation of the proposed messaging transaction to each destination number may be performed. Validation proceeds as follows. Subscriber 212 seeks to send a message to caller 210. Subscriber 212 enters caller 210's telephone number into a message delivery list ("MDL"). VMS 246 receives subscriber 212's entered list, including caller 210's telephone number. VMS 246 queries RMD 242, providing the caller 210's telephone number, as well as information identifying the subscriber 212. The RMD 242 uses the Tables I and II described above to determine the identity of the service provider serving caller 210 and the geographic location of both caller and subscriber.

Detailed Description Text (58):
With the above-described information concerning caller 210 and subscriber 212, RMD 242 is able to apply the rules set forth in the tables within RMD 242 to determine whether a message from subscriber 212 would be allowed. For instance, suppose subscriber 212 were located in Georgia and caller 210 located in Louisiana. Table II informs RMD 242 that an interstate message transfer between VMSs located in Georgia and Louisiana is allowed. The next step is to determine whether the VMSs involved different service providers. Table II informs RMD 242 that subscriber 212's VMS2 is operated by BellSouth and that caller 210's VMS1 by Evelyn's Voice Mail. With that information, RMD 242 examines Table III to determine whether Evelyn's Voice Mail accepts message traffic from BellSouth. It does, so the transaction is validated.

Detailed Description Text (59):
Each number of a regional messaging service subscriber that is entered into the MDL is verified through a similar process. If RMDs 240, 242 determine that state regulations do not allow interstate message traffic (e.g., if a participant in the proposed messaging transaction is in Alabama), the telephone number will not be validated. Or, in another example, if the message transaction is only inter-LATA, it would be allowed in North Carolina, but not Alabama. In yet another example, if the message transaction involves exchange of messages within North Carolina's "424" LATA, the method of the present invention determines such an exchange is allowed by regulation. But if the same transaction involves a recipient whose service providers were, for instance, Harry's VoiceMail, RMD 240, 242 determines from Table III that message exchange with Harry's VoiceMail is not allowed.

Detailed Description Text (60):
Once validation is complete, the subscriber 212 is alerted to the validation by the spoken name confirmation process described below. If a recipient is a non-subscriber 42, an announcement will so indicate, including the non-subscriber's 42 phone number. Thereafter the subscriber 212's VMS may simply store that number for later outdialing and playing of the message, as described below. Typically, a confirmation spoken name may be given for subscriber's 212 and a confirmation telephone number for non-subscribers 3.

Detailed Description Text (61):
When a distribution list is completed and the subscriber 212 orders the message circulated, messages to subscribers 212 are formulated into the VPIM format and sent via the TCP/IP network 252 described above. Messages to non-subscribers 3 will be "outdialed." In other words, messages will be forwarded to the subscriber's 212 local switch-or SSP, such as SSP 222. There, the message encounters a trigger (e.g., a TAT or Terminating Attempt Trigger) that makes the SSP 222 query the SCP 228 for the destination of non-subscriber 42 indicated in the list. This is a billing mechanism that is also used for flexible call forwarding features that is described in U.S. Pat. No. 5,991,377 that is owned by the assignee of the present invention and which is hereby incorporated in its entirety by this reference. SSP 222 thereafter dials an actual call, connecting the subscriber's VMS to the non-subscriber's voice mailbox so that the recorded message may be played and recorded in the non-subscriber's voice mail. In order to be compatible with caller ID services, the calling number field will display an appropriate name, like "[Name of the Messaging Service Provider] Messaging." Alternatively, the messaging service provider may itself route the call, selecting the route and carrier. Note that because non-subscriber messages are outdialed via a normal telecommunications call, they are not subject to the regulatory and business constraints for messaging traffic and will not need to be validated against those conditions.

Detailed Description Text (63):
In FIG. 3, a caller 210 sends a message to a subscriber 212. Caller 210 either calls subscriber 212's voice mailbox to leave a message or caller 210 originates a message in VMS 244 that ultimately is intended for subscriber 212's voice mailbox on VMS 246. Subscriber 212 receives the message and desires to formulate a reply to the caller 210.

Detailed Description Text (64):
Replies to Other Subscribers:

Detailed Description Text (65):
While the subscriber 212 listens to caller 210's message, VMS 244 queries SCP 226 to determine whether the caller 210 receives voice mail services from a service provider participating in regional messaging system 200. The VMS 244 query sends to the SCP 226 the caller 210's telephone number and the subscriber 212's telephone number. VMS 244, of course, stores subscriber 212's telephone number.

Detailed Description Text (66):
The present invention, using the process and system described above, will discern the service provider, LATA, and State of each party to the proposed transaction. Thus, the subscriber 212's VMS 246 queries the RMD 242 using LDAP queries forwarded over the VMS network 252 to ascertain the location of the VMS operating the recipient's voice mail service and the service provider operating that VMS. RMD 240, 242 will then apply the applicable regulatory constraints or business constraints stored in various tables, such as tables I through III above, to determine the validity of going forward with the proposed reply transaction.

Detailed Description Text (68):
Replies to Non-Subscribers:

Detailed Description Text (69):
Replies to non-subscribers need not be validated on regulatory and business agreement grounds, although there will have been an earlier validation query that confirms the destination is not to a subscriber. In that case, RMDs 240, 242 need not be further queried to determine the destination of the reply.

Detailed Description Text (70):

Instead, the subscriber 212's VMS 246 simply stores calling line identification ("CLID") information available when the non-subscriber calls to leave a message in subscriber 212's voice mailbox. If the CLID information is not designated as private, VMS 246 will play the telephone number as confirmation of the non-subscriber recipient. In any event, either after or during composition of the message, VMS 246 causes an actual call to be outdialed to the non-subscriber, as described above, through the subscriber 212's serving SSP 222 directly to the designated number so the message may be left.

Detailed Description Text (72):
Accurate delivery of messages is important to voice mail subscribers 200, who do not wish possibly confidential messages to be misrouted. To decrease subscriber 212 error in identifying message recipients, when a subscriber 212 chooses the reply option, the subscriber's VMS 246 plays a confirmation of the reply destination, called the confirmation name announcement.

Detailed Description Text (73):
If subscriber 212 chooses to reply to a message from caller 210 by entering the appropriate code or otherwise indicating that choice, VMS 246 queries the SCP 228 to retrieve the original caller 210's spoken name. Assuming caller 210 subscribers to system 200, SCP 228, in turn, accesses the RMD 242 to retrieve either the caller 210's spoken name or a pointer that identifies to VMS 246 the appropriate VMS messaging platform (e.g., VMS 244) to query for that spoken name. If the spoken name resides on a different VMS, VMS 246 is so informed by a reply LDAP message from SCP 226. Thereafter, VMS 244 launches another query to VMS 246 to retrieve the caller 210's spoken name. The spoken name is played to the subscriber 212 as confirmation that the reply will be forwarded to the intended recipient. Upon hearing the name, the subscriber 212 records the reply message. When the reply message is complete, the subscriber 212 so signals VMS 246 via entry of the appropriate codes. VMS 246 packages the reply and routes it back to the caller 210's VMS 244 using the information retrieved via SCP 226 from RMD 228.

Detailed Description Text (74):
If the calling name is unavailable, the messaging system 200 should play the calling number, if available, or other appropriate announcement to confirm the subscriber 212's choice. For instance, a text-to-speech rendering may be made of the caller 210's text-stored name or the originating number of the caller 210 may be provided to the subscriber 212 in order to inform the subscriber 212 of the identity of the message's recipient, caller 210.

Detailed Description Text (76):
The preferred embodiment of the regional messaging system 200 described above for use with the present invention supports the use of messaging to "sub-mailboxes." Sub-mailboxes are multiple mailboxes that can be accessed by dialing a single telephone number and then selecting a code that identifies a particular mailbox. For instance, a sub-mailbox may be identified by an additional number (00, 01, 02, etc.) associated with the main telephone number. Users control uses of submailboxes and may change which boxes are used for which users. If a subscriber attempts to send a message to a submailbox (either through the reply or message distribution options), the LDAP query process described above will be used to determine whether the submailbox exists. Subscribers will be informed if that mailbox is invalid.

Detailed Description Text (77):
The preferred regional messaging system 200 in which the present invention may be deployed accepts facsimile messages and stores them in a particular subscriber's VMS 244, 246. Facsimile data in certain protocols (like Group 3 fax messages) can be stored, forwarded to an appropriate printer or computer for written display or annotated. Facsimile messages may be forwarded like other messages using the same general process described above. When the subscriber 212 chooses a forwarding address involving a toll payment, the procedure described above may first be used to determine the location of the VMS serving that number. Other databases deployed at the RMD 240, 242 or the subscriber's VMS 244, 246 may determine whether that location will incur toll charges and an appropriate announcement may be played so notifying the subscriber 212.

Detailed Description Text (78):
Subscribers, in setting up their accounts, may be offered the option of associating a facsimile number with the subscriber's mailbox number. This allows such subscribers to receive voice replies to facsimile messages. Those replies may be made by generally following the same

process used to determine if a reply to a voice message may be made. Thus, the VMS 244, 246 determines whether the RMD 240, 242 lists the calling number associated with the facsimile number that was appended to the facsimile message left in the subscriber's voice mail. If there is an associated calling number, the subscriber retrieving or re-directing a facsimile message is offered the option of voice replying to the originator's mailbox.

Detailed Description Text (79):
RMDs 240, 242 may be implemented so that, as part of the profile information, it stores the spoken name of each subscriber to affiliated providers' services. In that case, the third party service providers may accept messages from the regional messaging system 200 by simply configuring their VMS to process messages that are in the appropriate format, such as the VPIM format described above. In other words, storing the spoken name at RMDs 240, 242 allows providers to more easily configure their voice mail systems to accept messages generated by system 200.

Detailed Description Text (81):
The present invention can also be deployed over other existing network, including a TCP/IP network. For instance, the present invention can be deployed over the internet. That can be accomplished by having users associate with their phone numbers their e-mail addresses. When a particular user desires to forward a message to another user part of the regional messaging system, the user's messaging server can utilize the process of the present invention to launch queries to the directory to determine whether or not the recipient is also participating in the region-wide messaging system. By retrieving the recipient's IP address (e.g., e-mail address), the present invention can thereafter formulate the originating subscriber's message and send it as a file attachment via e-mail to the message recipient. The e-mail attachment can be either an audio link or a transcript of the originating party's voice message. Such a transcript can be prepared using commercially available voice-to-text conversion software, which can be deployed on the same platform that holds the directory information.

Detailed Description Paragraph Table (1):
Description/Directory Field LDAP DN Attribute Subscriber's Mailbox Number CN (Common Name) Name Announcement Spoken Name MDSBlocking N/A

Detailed Description Paragraph Table (3):
Subscriber Table I Subscriber Identifier VMS Identifier Block Flags 770-555-1212 6110.atlngamm62 Subscriber in default; [e.g., VMS1 run by subscriber language BellSouth] features. 404-555-1212-02 2113.atlngaev63 [e.g., VMS2 Subscriber authorized run by Evelyn's Voice Mail] for messaging 336-555-1212 4331.atlngahh69 [e.g., VMS3 Subscriber paid up run by Harry's VoiceMail] and authorized

Other Reference Publication (5):
"General Recommendations on Telephone Switching and Signalling--Introduction to Intelligent Network Capability Set 1", International Telecommunication Union, XP-002141945, Mar. 1993.

CLAIMS:

3. A method according to claim 2 wherein the querying is performed by providing the messaging directory with at least the message recipient's telephone number correlated to at least the identities of one of the first or second voice mail servers.

14. A method according to claim 8 further comprising out-dialing a telephone call to a person not subscribing to regional messaging services in order to deliver a message.

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DOCUMENT-IDENTIFIER: US 6498791 B2
TITLE: Systems and methods for multiple mode voice and data communications using intelligently bridged TDM and packet buses and methods for performing telephony and data functions using the same

Abstract Text (1):
Systems and methods by which voice/data communications may occur in multiple modes/protocols are disclosed. In particular, systems and methods are provided for multiple native mode/protocol voice and data transmissions and receptions with a computing system having a multi-bus structure, including, for example, a TDM bus and a packet bus, and multi-protocol framing engines. Such systems preferably include subsystem functions such as PBX, voice mail and other telephony functions, LAN hub and data router. In preferred embodiments, a TDM bus and a packet bus are intelligently bridged and managed, thereby enabling such multiple mode/protocol voice and data transmissions to be intelligently managed and controlled with a single, integrated system. A computer or other processor includes a local area network controller, which provides routing and hub(s) for one or more packet networks. The computer also is coupled to a buffer/framer, which serves to frame/deframe data to/from the computer from TDM bus. The buffer/framer includes a plurality of framer/deframer engines, supporting, for example, ATM and HDLC framing/deframing. The buffer/framer is coupled to the TDM bus by way of a switch/multiplexer, which includes the capability to intelligently map data traffic between the buffer/framer and the TDM bus to various slots of the TDM frames. Preferably, a DSP pool is coupled to buffer/framer in a manner to provide various signal processing and telecommunications support, such as dial tone generation, DTMF detection and the like. The TDM bus is coupled to a various line/station cards, serving to interface the TDM bus with telephone, facsimiles and other telecommunication devices, and also with a various digital and/or analog WAN network services.

Brief Summary Text (4):
Businesses, particularly small to medium size offices, typically have a need for a variety of voice and data communications. For example, a typical office might have a dedicated fax machine, using a dedicated or shared telephone line, one or more telephone lines for voice communications, perhaps coupled to a central or distributed voice mail system(s), and one or more computers or computer networks, often coupled to telephone lines via one or more modems. Many offices now use the Internet in some form for business communications or research or the like, often by way of a modem or modem pool coupled to individual computers.

Brief Summary Text (6):
FIG. 1 illustrates a conventional small office communication configuration. Voice communication system 1 typically is implemented by way of multiple analog trunks 16 from wide area network ("WAN") 18. WAN 18 often consists of a telecommunication network by way of a local telephone company or other telecommunications service provider. Analog trunks 16 may be directed through switching system 10, which may be a conventional PBX or similar telephone switch. Telephones 12 and voice mail system 14 are coupled to switching system 10. Often, dedicated analog line 16A is coupled to facsimile 44 for facsimile communications.

Brief Summary Text (8):
Such a conventional system often is characterized by piecemeal equipment and network solutions, limited or non-existent coordination and management between voice system 1 and data system 2, non-optimized or non-integrated equipment, and inefficient use of costly network services (telephone lines, data lines, etc.), such as duplicate and often idle phone and data network lines, often provided from multiple equipment/service providers. In general, such conventional systems are neither constructed nor operated in a manner to provide efficient and integrated

voice/data communications.

Brief Summary Text (10):
The present invention is intended to address various disadvantages of such conventional communication systems. The present invention provides various systems and methods, perhaps more succinctly a platform, by which voice and data communications may occur in multiple modes and various protocols, and more particularly systems and methods for multiple native mode voice and data transmissions and receptions with a communications/computing system having a multi-bus structure, including, for example, a TDM bus, a packet bus and a control bus, and multi-protocol framing engines, preferably including subsystem functions such as PBX, voice mail and other telephony functions, email and/or file server, Internet server, LAN hub and data router. With the present invention, a platform and various processes are provided in which a TDM bus and a packet bus are intelligently bridged and managed, thereby enabling such multiple mode/protocol voice and data transmissions to be intelligently managed and controlled with a single, integrated system.

Brief Summary Text (11):
In preferred embodiments, a computer or other processor includes a local area network controller, which provides routing and hubs and/or switches for one or more packet networks. The computer also is coupled to a multiple buffer/framer, which serves to frame/deframe data to/from the computer from TDM bus. The buffer/framer includes a plurality of framer/deframer engines, supporting, for example, ATM and HDLC framing/deframing, and raw buffering of voice data or the like. The buffer/framer is coupled to the TDM bus by way of a multiple port or multiport switch/multiplexer, which includes the capability to intelligently map data traffic between the buffer/framer and the TDM bus to various slots of the TDM frames. Preferably, a DSP pool is coupled to one or more switch/multiplexer ports and/or the buffer/framer in a manner to provide various signal processing and telecommunications support, such as dial tone generation, DTMF detection and the like. The TDM bus is coupled to a various line/station cards, serving to interface the TDM bus with telephone, facsimiles and other telecommunication devices, and also with a various digital and/or analog WAN network services. The present invention provides a platform by which processing functions may be switched to provide support for a wide range of network, vendor and application services.

Detailed Description Text (5):
It will be appreciated that communications system 50 also may implement hardware and software for additional network functions, which are included in alternative embodiments. Such network functions include, but are not limited to: name server, such as DNS (Domain Naming System, which is used in the Internet for translating names of host computers into addresses) or WINS (Windows Internet Name Service, which is a name resolution service that maps or resolves Windows networking computer names to IP addresses particularly in a routed environment); firewall (as is known in the art, a firewall is a hardware/software implement that limits the exposure of a computing system such as communications system 50 or computers coupled thereto to access from a computer external to the system, which may include a network level firewall or packet filter that examines data traffic at the network protocol packet level, or an application-level firewall that examines data traffic at the application level, such as FTP or file transfer protocol, email, etc.); proxy server (as is known in the art, a proxy server is a type of firewall that uses a process known as address translation to map internal user IP addresses to the IP address associated with the proxy server firewall in order to provide extra security, etc.); DHCP (Dynamic Host Configuration Protocol, which is a protocol which allows a server to assign dynamically IP addresses to particular computers in real time, etc., which may support manual, automatic and/or dynamic address assignment, which may be used to verify a particular computer's identify, temporarily assign it an IP address for a particular period of time, and reclaim the IP address later for reassignment at the expiration of the particular period of time, etc.); and/or email server or gateway (which, as is known in the art, may be used to send and receive emails and/or send and receive faxes for the computers connected to the LAN or LANs, etc.).

Detailed Description Text (6):
Communications system 50 includes the functionality of what is known as a PBX (as will be described further). In preferred embodiments, communications system 50 is connected to a plurality of telecommunication devices, such as telephones 12, facsimile 44 and other suitable telecommunications devices and access and server functions (such as private voice mail, recording devices, WAN service interface cards, etc.). What is important is that communications

system 50 include interfaces for a plurality of telecommunications devices for the particular and complete office/work environment and infrastructure.

Detailed Description Text (7):
Communications system 50 is coupled to WAN voice/data services network(s) 58 through trunks 54. Voice/data services network(s) may include private line, local or long distance carrier networks, Internet, intranet and/or any other current or future WAN-type network services. Trunks 54 may consist of high, medium or low speed digital and/or analog lines, either public or private, and in certain preferred embodiments consist of high speed dedicated resources such as what are known as T-1, PRI (Primary Rate ISDN), ATM, VDSL, HDSL, ADSL, DDS (Dataphone Digital Service, also called Digital Data System), wireless, cascade, proprietary and/or twisted pair analog lines from a local telephone company. What is important is the communications system 50 is coupled to WAN services, trunks and the like in a manner that the user, service provider, administrator and/or algorithm has determined will provide adequate or required resources, on a cost-effective basis, for the particular office/work environment and operating conditions.

Detailed Description Text (10):
Communications system 50 is controlled by host processor/system resources 70, which in preferred embodiments include a computer powered, for example, by a commercially available or other microprocessor and an embedded and/or commercially available operating system. What is important is that processor/system resources 70 provide sufficient processing power, memory and storage resources (RAM, ROM, hard disk, magnetic or other storage, etc.), bus and other resources in order to control the various subsystems and components as will be described. In particular, computer/system resources 70 enables automatic internal negotiation, control and enabling of services and applications. Although not expressly shown, processor/system resources 70 also may include other components of a relatively high-end personal computer, workstation or server, such as a display device, keyboard, serial ports, parallel ports, power supply and the like. The various subsystems and components of communications system 50 are intelligently controlled, managed and monitored by processor/system resources 70. Processor/system resources 70 provides system and server management software and the like, and a platform for various server applications as described herein.

Detailed Description Text (12):
Processor/resources 70 also may be connected to DSP 76. DSP 76 preferably consists of a single digital signal processor or multi-digital signal processor resource pool, which serves to provide a variety of functions within communications system 50. In preferred embodiments, DSP 76 generates dial tones (such as for telephones 12), DTMF digit detection and decoding, echo cancellation, coding/decoding functions, voice conferencing, voice compression, voice recognition and the like. In other embodiments, DSP 76 performs data compression, transcoding, processing for voice communications using an Internet protocol ("IP") or other voice over other network protocol or the like. In general, DSP 76 provides a set of processing and memory resources to support the various voice/data services controlled and managed by processor/resources 70. As illustrated by bus connection 84A, DSP 76 alternatively may be coupled directly to TDM bus 78.

Detailed Description Text (14):
Coupled to TDM bus 78 are line, station, trunk, or other interface cards 82. Cards 82 provide CODEC, line interface, off-hook detect and other functions as are known in the art to support various telecommunication devices (such as telephones 12 and facsimile 44) and WAN-type network services (such as voice/data services 58) that are communicating with communications system 50 via TDM bus 78. In preferred embodiments cards 82 provide points of termination for a plurality of telephones 12, one or more facsimiles 44, and various T-1, PRI, ATM, analog and/or other WAN-type network services as determined by the particular office/work environment. Cards 92, under control of processor/system resources 70, may include points of termination for emergency or backup telephone services and the like, such as in the event of a power failure or to provide analog services in the event a dedicated resource such as a T-1 is unavailable for some reason.

Detailed Description Text (30):
Server encryption applications 23 may be provided in order to provide encryption or similar coding or processing of voice/data communications processed by communications system 50. VoIP gatekeeper 25 may be provided to service and control voice over Internet protocol ("VoIP")

communications. As more specifically described below, various types of VoIP communications may be effectively managed and controlled in accordance with preferred embodiments of the present invention, such as, for example, a determination that acceptable conditions exist on the Internet for such communications. Directory 27 may be provided in order to make various types of directory information available to users of communications system 50. Directory information provided by directory 27 may include names, telephone extensions, address or other personal or work information regarding persons or departments, etc., serviced by communications system 50. Directory 27 also may include similar directory type information for persons or departments, etc. in a remote or other locations, such as may be accessed through voice/data services 58.

Detailed Description Text (36):
For example, data switching services may be provided such as by LAN/NDIS/DDI drivers 39 (LAN, NDIS and DDI being exemplary) through hardware modules such as switched Ethernet 45 and hub 47. Routing services may be provided such as through WAN drivers (specific network services such as PRI and T-1 being exemplary) through hardware modules such as T-1 module(s) 49, ISDN module(s) 51, central office-plain old telephone service (CO-POTS) module(s) 53, V.35 module(s) (it should be understood that various hardware modules may be utilized in accordance with preferred embodiments of the present invention, as desired to implement the various data switching, routing and other communications connections as may be determined by the needs of the particular office/work environment. PBX station services, such as automated attendant, reception, voice mail and the like, may be provided through station manager 43. Station manager 43 provides hardware for connection to various telecommunications devices, such as phones 12, facsimile 44, etc. In general, station manager 43 provides sufficient interface hardware in order to connect to the various devices that may be determined by the needs of the particular office/work environment).

Detailed Description Text (37):
Additional features particularly of hardware components of such embodiments involving detection operations incorporating or utilizing DSP resources such as are included in preferred embodiments will now be described (DSP resources included in such embodiments are described, for example, in connection with FIG. 3. A technique for determining characteristics of an analog line is to send a known signal (preferably a known tone or combination of tones or frequencies of known energy, etc.) down a line, and convert a predetermined frequency (or frequencies) of a returned signal from the analog line to a voltage or to otherwise process the returned signal; characteristics of the analog are determined based on the voltage or otherwise from information extracted from the returned signal. In preferred embodiments the returned signal is processed by DSP resources (see DSP 76 of FIG. 3) in order, for example, to perform a Fast Fourier Transform ("FFT") or other signal processed on the returned signal. As example, particular frequency bands in the returned signal could be evaluated to determine whether a phone is physically connected to the line (e.g., an analog phone typically presents a 10K ohm impedance to the line in an on-hook condition, the presence of which could be determined by evaluation of the returned signal. In preferred embodiments, DSP resources could evaluate the returned signal energy, again preferably with an FFT, and the presence and/or type of telephone device physically attached to the line could be assessed/determined, and still preferably an assessment of the quality of the particular line could be made based on such an analysis of the returned signal.

Detailed Description Text (38):
Such signal processing could be done periodically or upon detection of errors, start-up or reboot, or upon initiation of a diagnostic or maintenance routine. With remote administration and configuration capabilities as described elsewhere herein, such phone presence detection, line quality assessment, etc., could be conducted from a remote location (such as enabling a central system administration to "map" the presence of phones to particular lines in a remotely located system. In accordance with such embodiments, such capability enables a similar functionality to the link status indicators that may be available on network ports. Such link status information for analog telephones can be incorporated into a visual representation of the system, easily viewable remotely via an HTTP link over the Internet, for example (such remote viewing of the physical status of a system, i.e., "chassis view," is described elsewhere herein). It should be understood that this approach to obtaining line status and information can easily be applied to other aspects of telephone lines. For example, the line condition, or suitability for high speed data transfer, or perhaps the highest speed available on a particular line (e.g., "speed grading" or "speed characterization" of individual lines) can be measured.

Detailed Description Text (43):
In the event a user picks up one of telephones 12, an off-hook condition is detected by the appropriate card 82, which signals processor/system resources 70 of the off-condition. Processor/system resources 70 controls switch/multiplexer 74 to couple the appropriate card 82 to DSP 76, which generates a dial tone that is coupled to the appropriate telephone 12. The user hears the dial tone and may then proceed to place the desired call. DSP 76 detects the digits of the telephone number of the desired call and provides the detected digits to processor/system resources 70. For an internal call, processor/system resources 70 directs that the called internal telephone receive a ring signal from the appropriate card 82. Upon pick-up of the called internal telephone, the telephone connection between the internal phones is established by way of TDM bus 78 and the appropriate cards 82.

Detailed Description Text (45):
Incoming calls are detected by the appropriate cards 82 and signaled to processor/system resources 70. Connections of voice incoming calls to telephones 12 are established under control of processor/system resources 70 over TDM bus 78.

Detailed Description Text (49):
In accordance with preferred embodiments of the present invention, one or more of computers 24 may execute a PBX/telephony control application software program. In accordance with the PBX/telephony control application, hereinafter referred to as the "office attendant type" program, control of the telephony and related functions of communications system 50 may be intelligently managed and controlled. With such an arrangement, one or more computers on the LAN may be used to control incoming and outgoing calls of the office using the computer in a natural and intuitive manner. A telephony headset or telephone preferably is associated with the particular computer that will be running the office attendant type program to enable traditional voice communications with incoming callers, etc.

Detailed Description Text (50):
As illustrated in FIG. 6, a party desiring to control the incoming and outgoing calls and/or station to station calls of the office ("attendant 1") may log-on and run the office attendant type program from one of the computers connected to the LAN connected to communications system 50. At step 100, attendant 1 may be required to enter an appropriate user name/ID and password in order to recognize attendant 1 as an appropriate user to assume control of the telephony functions of the office. A network or systems administrator may set up password control for parties authorized to run the office attendant type program. At step 102, in preferred embodiments the computer running office attendant type program has downloaded to it the current telephone subscriber directory such as over packet bus 80A or 80B of FIG. 3 (e.g.: a complete listing of the telephone subscribers; extensions; status information such as do not disturb, forward and forwarding information, forward to voice mail, hunt group information, etc.) from communications system 50. In this manner, the computer or computers running the office attendant type program may locally contain current subscriber information for controlling the incoming and outgoing calls of the office. In preferred embodiments, communications system 50 automatically determines when subscriber information changes, e.g., a subscriber has been added to or deleted from the telephone directory, or an extension has changed, or a subscriber's status information has changed, or any state associated with communications system 50, etc., in order that updates may be timely made available. In such embodiments, computers running the office attendant type program may be updated promptly and automatically by communications system 50 so as to contain current subscriber information on an ongoing basis to more efficiently control telephony operations of the office. It also should be noted that in preferred embodiments the subscriber information also may include other information, such as the email address and extended directory information including personal information manager ("PIM") information of the particular subscriber and network identification for a computer associated with the particular subscriber. With such information, net-messages or other communications with particular subscribers may be facilitated as more fully described herein.

Detailed Description Text (51):
It also should be noted that this subscriber download concept is applicable in various forms to all computers coupled to communications system 50. For example, communications system 50 includes information regarding all users registered in the PBX (i.e., all users having a telephone extension and/or computer coupled to communications system 50 such as over the LAN or WAN). Thus, in the event of a subscriber directory change, communications system 50 may

"broadcast" updated subscriber directory information to all computers coupled to communications system 50, or, in alternate embodiments, communications system 50 sends a net message, email or other message to such computers coupled to communications system 50 that prompts the users of such computers to the availability of the subscriber directory update (e.g., the remote computers received a message indicting the availability of the subscriber directory update, which preferably includes an "accept" icon and a "reject" icon, thereby enabling the user to receive or not receive the update as he/she may desire).

Detailed Description Text (58):
Referring to FIG. 22, the Main Window preferably includes a small appearance GUI footprint including three low profile line status indicators. Office communicator-type programs preferably do not include a `Calls in Queue` or a `Calls on Hold` indicator. Alternative views of this window can be sized and displayed to take up less physical space on the screen for the end user. Such feature buttons allow additional functionality to be added into the program, for example, multiple call parking features can be added. In this example, there are two types of park: Self-Park and System Park. Self-park preferably parks the call at the extension of the person parking the call. Hence, if an outside caller calls extension x125 and the user at x125 answers and self parks the call, then the user at x125 can page and announce "Pick up x125". System park returns a parking address, or slot number of a predetermined number of parking spaces that the system allocates for such call parking. Hence, if an outside caller calls extension 125 and the user at x125 system parks the call, then the display on ext 125's office communicator-type program will read: "Call Parked on <slot number>", e.g. "Call Parked on 2". Then the user at x125 can page, and announce "Pick up 2".

Detailed Description Text (59):
Referring now to FIG. 23, such an office communicator type program that is optimized for general telephone and computer use, can include a screen pop window as illustrated. The main user interface illustrated in part in FIG. 22 preferably consists of a three-line display. However, this main user interface is not intended to be maximized at all times. When an incoming call arrives, the screen pop illustrated in part in FIG. 23 will slide out and occupy a small portion of the screen to let the user know that there is an incoming call, and provide caller information to the user. In addition, such a screen pop may incorporate a visual signal, e.g., a rotating telephone icon, to help indicate that a call is trying to get through. When there are new messages at the extension, the screen pop will also appear to indicate (via an appropriate icon or other indicia, preferably rotating or otherwise moving in order to attract visual attention, etc.) that there is a message waiting. For making outbound call and other simple/more frequent call control operation, a toolbar with basic call control functions preferably is provided to the user. Other visual and operational variations suitable for other working environments will be apparent from the above discussion.

Detailed Description Text (64):
The user preferably may also initiate a call from the phone. The user would pick up the phone and hear dial tone. He or she can then dial the number from the phone set. When the user is already on another call and he wants to make another call by the application, he can choose to put the current caller on hold and dial the number, or the application would automatically put the current caller on hold when he dials the number. When the user is already on another call and he wants to make another call by the phone, he can put the current caller on hold by hitting `FLASH` on the phone and dial the number.

Detailed Description Text (65):
The user can put a current call on hold using the mouse, by using the keyboard or by using the phone. By making an outbound call, or answering another call from the application, the current call can automatically be put on hold by the application. The user can put the current call on hold from the phone, for example, by hitting `FLASH` on the phone set.

Detailed Description Text (74):
During the transfer of a call, if the destination extension is on the phone or on DND (Do Not Disturb), the application preferably presents 3 options to the user. The user can put the caller on hold, sent the caller to the voicemail of the destination, or send a NetMessage to the destination's computer. On the receiving end of the NetMessage, the user would see a dialog box on his machine with the text message and 2 options i.e. accepting the call or ignore the call. If the user chooses to accept the call, the call automatically transfers from the originated extension to the destination. If the user chooses to refuse the call, the

application will notify the originated user that the call was refused.

Detailed Description Text (88):
Referring now to FIGS. 8A to 8D, exemplary windows from illustrative preferred embodiments of office attendant-type programs in accordance with the present invention will now be described. As illustrated in FIG. 8A window 130 includes one or more line displays 132 (five are shown in FIG. 8A for illustrative purposes) for indicating various telephone lines available in the particular application of communications system 50. The number of telephone lines, of course, may be tailored for the particular application. Preferably positioned adjacent to line displays 132 is call/line status display 148 for displaying symbols adjacent to each line indicative of the status of the line, such as idle, phone ringing, active call in progress, call on hold, hold recall alert, etc. Status display 148 provides a ready visual indicator to the user of the office attendant-type program of the status of the various telephone lines that are being monitored. Also adjacent to the line displays (as illustrated adjacent to status display 148) are user identification displays 150, which serves to display the name and/or extension or telephone number of one or both parties to a call. In certain embodiments, caller ID type information may be obtained by communications system 50 from an appropriate interface card (see interface cards 82 of FIG. 3) and also displayed on displays 150. Displays 150 also may display a clock indicating the duration of a call on a particular line.

Detailed Description Text (95):
In preferred embodiments, calls may be directed to the computer running the office attendant type program because a main number has been directed to this computer (and its associated telephone or headset), or because calls have been forwarded to the office attendant type program, or because a called party is on the phone, has indicated the called extension is "do not disturb," etc. In such situations, the office attendant type program user may need to transfer calls to other extensions, either inside the office or outside the office.

Detailed Description Text (96):
Preferably, persons in the office have a computer running a program in companion with the office attendant-type program. Such windows may include, for example, an animated icon, caller ID information, etc., and may include one or more icon the clicking of which causes the call to be answered. In such preferred embodiments, the office attendant type program may cause one or more windows to appear on the computers of particular persons in the office, such as a person to whom a call is being directed. As an illustrative example, a call may come in through WAN services network 58 (see, e.g., FIG. 3) and be directed to a main telephone number, which may be designated to be forwarded to a telephone associated with a person running the office attendant type program on a particular computer 24, and may be so directed by way of TDM bus 78 and switch/multiplexer 74, under control of processor/system resources 70. The computer 24 running the office attendant type program may be used to transfer the incoming call to a particular extension, which may be readily accomplished by way of transfer icon 178 (see FIG. 8A).

Detailed Description Text (98):
In accordance with preferred embodiments of the present invention, in the event of a failed transfer, for example in case the extension to which the call is being transferred is busy, a window preferably is automatically displayed on the computer running the office attendant type program. An exemplary window 208 is illustrated in FIG. 9B. As illustrated, display 210 may display a descriptive message, such as "line busy," "do not disturb," etc. Preferably, a number of icons also are simultaneously displayed to aid the office attendant type program user in processing this call. Hold icon 212 may be used to place the caller on hold. Message icon 214 may be used to initiate a net message to the party to whom the call is to be transferred. Voice mail icon 216 may be used to direct the call into the voice mail of the party to whom the call was to be transferred. Cancel icon 218 may be used to cancel the transfer operation. With such an automatically generated window 208, the office attendant type program user is presented with options to more quickly process such calls, again preferably with a single or very few clicks of the mouse or pointer.

Detailed Description Text (99):
In certain embodiments, activation of hold icon 212 automatically "parks" the call on the extension of the party to whom the call is to be transferred. In certain embodiments, particular subscribers may have the option to program their extension so that calls parked on their extension may or may not be automatically connected once the called party has completed

its current call. In such embodiments, it may be desirable to have the called party informed that a call is being held. Preferably in such embodiments, the office attendant type program may be configured to automatically send a message (over a packet bus, as described earlier) to the computer of the party to whom the call is to be transferred, such as is illustrated by window 220 in FIG. 9C. In such embodiments, window 220 may contain message box 222, which may contain a message such as "call holding" or "call holding from Mike at extension 226," or "call holding; outside caller, number xxx," etc. What is important is that message box 222 display a message that a call is holding, with appropriate information identifying the caller displayed to the extent possible or desired. It should be noted that in certain embodiments caller ID information is displayed, and in some such embodiments a directory or library of names or other identifying information may be contained in communications system 50 and/or one or more of the computers connected to the LAN so that names or other identifying information may be associated with the caller ID information and displayed in message box 222. Preferably, the computer of the called party plays an audible tone or sound.

Detailed Description Text (102):
It should also be noted that, in the event of a particular user extension being dialed directly without going through the office attendant type program, a window such as window 220 of FIG. 9C may be displayed on the computer of the called party, either automatically for all calls, or only in the event that the called party has put his telephone on do not disturb, but has configured his extension to receive a message notification of calls, or in the event that the called party is on the line. In such embodiments, communications system 50 may generate such a window by a suitable message sent over by packet bus to the user's computer. In such embodiments, communications system 50 may simultaneously ring a user's extension and notify the user of the call with a net message, with the called being accepted, parked or forwarded to voice mail such as described earlier. Of course, in the event that a user previously configured his extension to be automatically forwarded to another extension or location or to voice mail or the like, then communications system preferably takes the programmed action directly. As an illustrative example, a user may configure his extension so as to route all calls to another extension or to a local or long distance telephone number. Such a user also may configure his extension so as to route all calls as voice over IP ("VoIP") call. In the later situation, processor/system resources 70 and/or DSP 76 may process the incoming voice information (received through the appropriate station card 82 and via TDM bus 78, etc.) into appropriate IP packets, which may then be routed, for example, through an HDLC framer/deframer 73B, through switch/multiplexer 74, over TDM bus 78 and out over a designated IP connection via WAN services 58, etc.

Detailed Description Text (103):
As previously described in connection with FIGS. 8A and 9B, a user running the office attendant type program preferably is presented with icon 174 (FIG. 8A) and icon 214 (FIG. 9B) for generating net messages, such as to send a net message to a user to whom a call is to be transferred, or to otherwise send a net message to a particular user, etc. FIG. 10A illustrates window 230 as an exemplary net message window that may be generated in response to clicking icon 174 or 214. As illustrated, window 230 preferably includes box 232 to identify the recipient of the intended net message, which may be automatically selected by the office attendant type program in the event of a failed call transfer situation. Otherwise, the recipient may be selected by pull-down menu as illustrated, or by direct entry of a name or extension number, etc. In preferred embodiments, as letters of the name is typed, the office attendant type program automatically scrolls through the subscriber directory in order to more arrive at the desired net message recipient.

Detailed Description Text (106):
In alternate embodiments, net messages may be sent from a computer running an office attendant-type program or a companion program, to any other computer coupled to communications system 50, either by way of the LAN or WAN, etc. In such embodiments, for example, if the user to whom a message is directed is logged onto communications system 50, the net message may be sent (preferably via communications system 50) either as a net message as previously described, or in the form of a visual "pink slip," "yellow sticky note," etc., which preferably appears in a small window on the screen of the user/message recipient. Still preferably, such "pink slip" or "yellow sticky note" messages include icons for options such as reply, delete, file/store, minimize, etc.; preferably, after a reply, delete, and/or file/store command, the message window automatically disappears. In certain embodiments, if a plurality of such messages are received and have not been processed so as to disappear, then such messages automatically stack

up, with a visual representation of stacked messages presented to the user (e.g., showing a third dimension of a stack of messages, etc.). In such embodiments, the user preferably sees the most recently received message on top, and also has the option to freeze/hold the updating of the message stack such as by selecting a suitable icon (e.g., if the user is reading a particular message, he/she may command that the message being read is not replaced by a subsequently received message), scroll through the stack of messages, etc. Still preferably, the user may select (again my suitable icon) that a particular message be forwarded to himself/herself as email, or to another person either as a similar message or email, etc. In preferred embodiments, communications system 50 automatically stores and sends as email all such messages that are not processed in a definitive manner by the user (e.g., if the user logs off without having replied, deleted, stored, etc. such messages, then communications system 50 processes such unclosed messages as emails to the particular user or users, etc.).

Detailed Description Text (109):
As indicated, conference icon 172 may be utilized to initiate a conference call in accordance with the present invention. Alternatively, in other preferred embodiments the conference call may be initiated by a click and drag operation. For example, an icon indicating a received call or the status of a received call (such as described earlier) may be clicked and dragged over the opened dialpad (see, e.g., FIG. 8A). The office attendant type program recognizes this click and drag operation as a request to open a suitable conference window, and the office attendant type program thereafter automatically opens the conference window.

Detailed Description Text (111):
In the event that icon 264 is selected, a call others operation may be initiated. FIG. 11B illustrates one embodiment of window 270 for calling additional attendees. As illustrated, window 270 preferably includes dialpad 272, which may be utilized to dial the extension or telephone number of a party to be added to the conference, which may be a party either on premises or off premises. Window 274 may be used to access either personal or system contact information, or both personal and system contact information, such as previously described. The names of particular subscribers may be entered or displayed in window 273, and the extension or number of a particular party to be added to the conference may be entered or displayed in window 276. Additional attendees may be added with icon 278 or removed with icon 280, with the additional attendees identified in window 282, with attendees in the conference identified in window 284. The next icon 286 preferably may be used to proceed to a dialog box from which the additional attendees may be called to join the conference. Selecting the finish icon 288 preferably results in the conference commencing or continuing without proceeding to a call dialog box.

Detailed Description Text (117):
It also should be noted that an office attendant-type program also may be run from a location remote from communications system 50, such as on a computer coupled to WAN services network 58 of FIG. 3. In such embodiments, a remote computer coupled to communications system 50 over a WAN network connection may run the office attendant-type program and remotely control the telephony functions of the office, in a manner such as described previously herein. Thus, control of telephony functions may be effectively performed in the office or remotely from the office, with control passed from computer to computer in an efficient and desired manner. Additionally, the user of the remote computer may run an office attendant-type program or a companion program as described elsewhere herein, and from such remote location be coupled to communications system 50 and remotely reconfigure the telephony and/or voice mail settings for the particular user. As an example, the remote user may use the remote computer in order to direct telephone calls to his/her extension to voice mail, or alternatively to have such calls forwarded to another extension or to a remote telephone number. With such embodiments, particular users may remotely access communications system 50 and, for example, control the forwarding of calls to an internal or remote location. As a particular example, a user using a notebook computer or PDA, etc., may couple to the Internet or WAN, etc. from a remote location, and direct that telephone calls to his/her office extension be forwarded in a desired manner (e.g., off-premise call forwarding, etc.). With the user able to access communications system 50 and remotely set and store PBX-type settings remotely, a variety of desired reconfiguration options are presented to the user.

Detailed Description Text (119):
In preferred embodiments, communications system 50 may dynamically associate physical telephones 12 with particular user extension numbers. In certain respect, this may be

considered like a "DHCP" (described elsewhere herein) for physical telephones. For example, a system administration may run a configuration/administration program (such as described elsewhere herein) and configure an extension number (e.g., 200) for a particular user, including associated parameters for such user, such as telephony and voice mail options (e.g., user forward settings, including off premise call forwarding, busy forward settings, ring-no-answer forward settings, time of day forward settings, display name for telephones displaying caller names, etc., whether the telephone is configured to be a telephone for a user running an office attendant-type program, etc.). At this time, the system administrator may or may not assign a physical telephone to that extension. Thereafter, the system administrator may notify the user that his/her extension number is 200. The system administrator also has the ability to enable and/or assign physical telephones. In the event that the system administrator has not assigned a physical telephone to that user, the user preferably has the ability to assign a physical telephone to his/her extension. For example, the user may pick up a telephone that has been enabled, and preferably does not have an extension assigned to that telephone, and the user enters a special code, e.g., numbers that communications system 50 recognizes as a request to assign a physical telephone. In certain embodiments, communications system 50 audibly informs (such as using DSP 76) the user of the status of that physical telephone (e.g., enabled or disabled, presently assigned to an extension, etc.). Thereafter, the user preferably is prompted audibly to enter his/her extension number. Optionally after a confirmation prompt, communications system 50 then assigns that physical telephone to the particular user. Still optionally, if the particular user extension is already assigned to another physical telephone, then communications system 50 un-assigns the other physical telephone at the time a new physical telephone is assigned to the particular user/user extension.

Detailed Description Text (120):
As will be appreciated, with such embodiments a special code also may be provided to un-assign physical telephones from particular user extensions, which preferably is implemented with password protection for particular users to ensure that the user's extension may not be assigned or re-assigned to physical telephones without the user's authorization or control (e.g., after entry of the extension number, communications system 50 prompts the user for a password associated with that user extension, and only allows assignment of a physical telephone to that extension if the correct password is entered, etc.). Thus, a user may assign his extension to a physical telephone by picking up that telephone and entering appropriate commands via the telephone keypad, and may un-assign his/her extension from that physical telephone by similarly picking up the physical telephone and entering appropriate commands via the telephone keypad (or by assigning the extension to a different physical telephone, as previously described), etc. In accordance with such embodiments, various office telephony arrangements may be implemented, such as an office arrangement in which a plurality of cubicles, offices or other physical spaces are provided with physical telephones but are not assigned to particular users. In accordance with such embodiments, particular users may be assigned an extension, and may occupy an available physical space and assign the physical telephone in that physical space with the user's extension. At the end of time for occupying that physical space, the user may un-assign his/her extension from that physical telephone, and then re-assign the extension to another physical telephone when the user later occupies another physical space, etc.

Detailed Description Text (121):
Additionally, as previously described communications system 50 may serve as an email server or otherwise serve to distribute email to particular computers (such as computers 24) coupled to communications system 50. Thus, communications system 50 can store information indicating that a particular user or users have received email. In such embodiments, communications system 50 preferably provides a visual or audio indication to the user that he/she has email. As illustrative examples, a special dial tone or message may be generated (such as with DSP 76) and presented to the user's telephone so that, when the user picks up his/her telephone, the special dial tone or message alerts the user that he/she has email (which also may include a special tone or message indicating that the user has voice mail). As one example, the tone or message may be a particular sound, but preferably is an audible message such as "you have email," or "you have voice mail and email" or "you have voice mail," etc. In the event that communications system 50 is implemented with telephones 12 having message indicator lamps, a particular lamp or blinking sequence may be used to indicate that the user has email, voice mail or both, etc. In all such embodiments, users may be desirably informed that they have email and/or voice mail with their telephony device (e.g., telephone).

Detailed Description Text (122):
As described elsewhere herein, communications system 50 may serve to provide email services to particular users with telephone extensions associated with communications system 50, etc. In addition, communication system 50 also provides a platform (such as with processor/system resources 70) on which various management, administration or other types of applications may be run (exemplary such applications are described elsewhere herein). In one embodiment, various WAN and other information is provided using an what is known as a SNMP-type protocol (as is known in the art, SNMP stands for Signaling Network Management Protocol, which is a protocol/method by which network management applications can query or request information from a management agent (such as are implemented in the present invention with processor/system resources 70 and appropriate software, etc.). A novel aspect of such embodiments of the present invention is that the voice mail system of communications system 50 also is implemented in a manner to provide voice mail related information in an SNMP-type form. Thus, in accordance with such embodiments of the present invention, communications system 50 stores a variety of information relating to voice mail, such as information relating to the status of the voice mail system, failure or alarm-type information, usage statistics, etc. In such embodiments, any tool or application that is SNMP compliant can access and view such voice-mail related information. Exemplary voice-mail-related information that may be made available via SNMP to an SNMP compliant tool or application is set forth in Table 1. With such embodiments, network (WAN and LAN, etc.) and PBX information along with voice mail-related information may be desirably provided using SNMP to a variety of SNMP tools and applications.

Detailed Description Text (130):
As also described elsewhere herein, in preferred embodiments VoIP communications may be readily enabled. Referring again to FIG. 3, voice from a telephone 12 may be coupled via station cards 82 and TDM bus 78 to switch/multiplexer 74. From switch/multiplexer 74, the voice data stream may be directed to DSP 76, which directly or in conjunction with processor/system resources 70, produce appropriate IP packet data (in effect, DSP 76 and/or processor/system resources 70 serve as, for example, a TCP/IP processor). After IP packeting, the voice data maybe directed to WAN services network 58 via an HDLC framer/deframer 73B (such as described elsewhere herein), or may be directed to one or more packet buses/LANs, also as previously described. It should be noted that, with DSP 76, which may be configured to provide substantial processing resources, voice data may be IP processed effectively with minimal or no consumption of the resources of computer/system resources 70, thereby helping to prevent an undesirable loading of computer/systems resources 70.

Detailed Description Text (148):
The multiple-bus architecture, application prioritization and isolation, and automatic route selection adds to the performance of communication system 50. These features ensure high-grade voice quality by keeping voice and data in their native environments, allow conversion between the voice and data environments to support services such as voice over IP (VoIP), maximize investment by making community resources, such as DSPs and WAN/LAN interfaces, available to both voice and data applications, keep mission-critical communications systems functioning under heavy load by ensuring they receive required system resources, provide flexibility in routing calls, and least-cost routing saves money by dynamically selecting trunks based on criteria selected.

Detailed Description Text (155):
The PBX capabilities will now be described. Communications system 50 PBX provides a full-featured, nonblocking digital PBX with sophisticated call control capabilities. These capabilities are delivered using standard analog telephones connected to your existing phone wiring. In addition, communications system 50 supports advanced call control capabilities over IP-based networks, for applications based on the Microsoft Telephony Application Programming Interface (TAPI) standard. TAPI allows the communication system 50 to optionally provide virtual digital telephones, delivering advanced call control features over inexpensive standard analog phones.

Detailed Description Text (161):
Communications system 50 PBX and Office attendant type program specifications are now shown below. PBX features for call features include the following: Call forwarding, Off-premise call forwarding, Transfer on busy and no answer, Time-of-day call forwarding, Call hold, Call toggle, Call waiting, Consultation call, Consultation transfer, Blind transfer, Conference call, Call pickup, Public address system support, and Do not disturb. The features for calling

and called party identification are as follows: support for enhanced caller ID phones, and Extension-to-extension identification.

Detailed Description Text (163):
The following are the office attendant type program features: (1) System--Standard Windows application; Call control over IP; Software-based console that is easy to relocate; Drag-and-drop dialing and conferencing; Virtual line appearances; Interface indicators signal call status; Caller ID display; Calls in queue display; Company telephone directory; Lookup-as-you-type dialing; Personal call log; Account number entry; Personal information manager; Conference manager; System speed-dial buttons; Programmable feature buttons; Most recently used numbers list; Login security; CTI link test button; Context-sensitive help; and Contact database importing; (2) Call handling--Dial pad; Hang up; Transfer with look-ahead; Hold; Answer next; Call forwarding; Do not disturb; (3) Installation requirements--66-MHz 486 PC with 16 MB of RAM (Pentium recommended); and Windows 95 or Windows NT 4.0.

Detailed Description Text (167):
The Communications system 50 AutoAttendant application economically processes inbound calls 24 hours a day--answering each call, providing customized instructions based on the time of day or day of week, and routing callers to the person best able to help them. Callers can use the intelligent call distribution feature to reach a particular person or department, without requiring an operator or direct inward dial (DID) services. For companies that use DID, AutoAttendant is ideally suited for assisting a live operator by handling common requests for information such as directions and mailing addresses.

Detailed Description Text (180):
Communications system 50 Remote Management System addresses these cost-of-ownership issues by providing integrated remote management capabilities for both voice and data services. Designed for remote management and fault monitoring, the Remote Management System provides a cost-effective method for managing the entire customer premise remotely. Companies with multiple offices or plans to expand can realize significant cost savings by leveraging their expensive technical resources, no matter where they are located. Furthermore, the centralized management capabilities of communications system 50 present a unique managed network service opportunity for both voice and data service providers.

Detailed Description Text (188):
Referring now to FIG. 14, additional preferred embodiments utilizing advanced call logging features will now be described. As illustrated in FIG. 14, call logging window 350 may be opened by a user of an office attendant-type program running on a computer in accordance with the present invention (see, e.g., FIG. 8A, call log icon 142). In alternative embodiments, call logging window 350 may be automatically opened upon receipt of an incoming call, or upon initiation of an outgoing call. Window 350 preferably includes display windows 352 and 354, which preferably displays information for calls in the log, such as a call log identification number, begin call time, end call time, duration of call, type of call (either inbound or outbound), account information, etc. In other embodiments, other information desired to be included in a call log record is included in such a window. Window 354 is illustrated with only one call displayed, although it should be understood that a plurality of calls my be displayed in window 354, and in fact the call log can include numerous calls that cannot be displayed simultaneously in window 354. A scroll button or buttons (such as scroll icon 353) preferably are provided to scroll up and/or down the logged calls.

Detailed Description Text (190):
In certain embodiments, upon receipt of an incoming call or upon initiation of an outgoing call, a window such as window 350 automatically appears (this may be by way of the office attendant-type program for a user who is managing incoming and outgoing calls of the office, or by way of a companion program for a user not managing incoming and outgoing calls of the office). In preferred embodiments, the user is prompted by a brief message displayed on the screen and/or an audio message played on the user's computer to enter the account number in window/field 358. In still other embodiments, the user must insert an account number in window/field 358 in order to complete the incoming or outgoing call. In such embodiments, processor/system resources 70 and/or the user's computer promptly reads any account number information provided by the user and any accepts or validates the account number (e.g., compares the entered account number to a stored list of valid account numbers, and determines if there is a match). In the event that an invalid account number is detected, a suitable

message window and/or audio alert indicating that the account number entered is invalid, unrecognized, etc., preferably is provided to the user. In the event that a valid account number is detected, then the call is completed.

Detailed Description Text (192):
In still alternate embodiments, communications system 50 (and/or another computer coupled to communications system 50 via a packet bus, etc.), periodically polls the computers utilizing a program with a call logging such as previously described retrieves the call log information. With automated call log polling, a central resource such as communications system 50 (and/or another computer) may periodically, and preferably automatically, collect call logging information over the packet bus (again, see, e.g., FIG. 3), which may be then made available to a suitable application running on communications system 50 and/or another computer, and compiled, processed, analyzed, printed, etc. In accordance with such embodiments, incoming and outgoing calls may be desirably logged and associated with account information, with such logged information desirably collected from a plurality of computers and made available to a central resource for further processing and/or use.

Detailed Description Text (221):
It should be noted that communications systems 50 illustrated in FIGS. 18 and 19, for example, also have coupled thereto a plurality of computers, telephones, etc., as previously described for purposes of generating, receiving various data streams, etc., although such details have not been shown for ease of description.

Detailed Description Text (222):
As described elsewhere herein, various voice mail type options may be presented to users of such communications systems in accordance with the present invention. One such advantageous voice mail option provided in accordance with preferred embodiments of the present invention include advanced email or voice mail-type broadcasts of desired messages. A user may decide to send a voice mail or email to some or all users of the communication system. With a suitable office attendant-type or companion-type program, for example, a user may select from a group list, etc., a desired group of persons to receive the communication. A broadcast voice mail, for example, could be input through the user's telephone in a conventional manner, and routed (see FIG. 3) through, for example, DSP 78 (via TDM bus 78, switch/multiplexer 74, etc.) which converts the voice mail message into an suitable data format, such as what is known as a WAV file, etc., and then sent via (for example) packet bus 80A and/or 80B to a plurality of computers. Communications system 50 also, for example, can record which users have received or not received the communication so that users may later receive the communication (such as when they log on at a later time). In addition, communications system 50 also has the capability to parallely process the communication as a message that is to be sent to persons via, for example the Internet. Using an HDLC framer/deframer as is provided in accordance with the present invention, a user may generate a voice mail or email communication that the communications system sends as packetized data over the LAN to recipients recognized to be users having a computer on the LAN, while generating a suitable HDLC, ATM framed communication to recipients who are reachable over the WAN, such as over the Internet or other IP connection, etc.

Detailed Description Text (225):
It also should be noted that, in preferred embodiments, DSP 76 is coupled to switch/multiplexer 74 in a manner so that it may "tap" into the various TDM data streams. This provides a significant improvement over systems in which data streams must be directed into a resource such as DSP 76, and then sent from DSP 76 over a separate channel, etc. (thereby utilized two channels, etc.). In such embodiments, DSP 76 can tap into or monitor data streams on particular TDM channels and provide, for example, processing to accomplish recognition (voice or speech, etc.), detection (such as of a fax or modem call, etc.), compression (including compression, transcoding, streaming and storing, etc.), packetizing (such as to prepare a data format such as for an email, etc.). In one illustrative example of such embodiments, communications system 50 may be programmed so that particular users (e.g., president, technical support, warranty claims line, etc.) automatically have voice mails stored as voice mails and also as an email or other data form. Thus, a voice call may be directed into voice mail, while DSP 76 concurrently processes the voice data stream into another form (e.g., email, data file, etc.), which may be stored, send over the WAN or LAN, etc. Having DSP 76, and particularly configured (such as with switch/multiplexer 74) so as to tap into the various channels, provides significant advantages in a variety of applications.

Detailed Description Text (228):
Backup communications module 416 preferably includes bus interface 420 for coupling information to/from bus 414, memory 424 for storing various information, as will described hereinafter, CPU 418, FLASH or other programmable memory 426, and modem or other communication unit 428. Module 416 preferably includes a standby or backup power supply 434, although in certain alternate embodiments communication unit 428 is coupled to, for example, link 430 of WAN services 58E, which may be a dedicated telephone line, POTS line, etc., which provides sufficient power to module 416 so that power supply 434 is not required. In such alternate embodiments, the various components of module 416 are implemented in low power CMOS technology or the like, and consume sufficiently low amounts of power so that module 434 may operate at a suitable speed in order to provide backup communications using only the power provided by link 430, such as, for example, in the event of a power failure in communications system 50 or the office in which communications system 50 is located, etc.

Detailed Description Text (233):
As previously described, module 416 preferably has a dedicate line (e.g., a POTS line) for such backup communications, and telephone 12 optionally may be coupled to such line for emergency voice calls or the like, etc. In alternate embodiments, however, communications unit is also (or alternatively) coupled to channels of TDM bus 78. In certain embodiments, a predetermined channel or channels of TDM bus 78 are dedicated for such backup communications. In other embodiments, communication unit 428 is coupled to TDM bus 78 through switch 432, and in such embodiments dedicated TDM channels are not required.

Detailed Description Text (236):
Referring now to FIGS. 25 to 35, exemplary embodiments of programmable digital telephones coupled to communications system 50 and an accompanying GUI configuration will be described.

Detailed Description Text (237):
FIGS. 25 to 29 illustrate preferred embodiments of the physical design of four programmable digital telephones. FIG. 25 illustrates 8-button telephone 466. FIG. 19 illustrates 8-button telephone 468 with display. FIG. 20 illustrates 16-button telephone 470 and FIG. 21 illustrates 64-button telephone 472. Preferably each digital telephone can be programmed individually through a series of GUI windows in the user configuration module, to which will be described below.

Detailed Description Text (238):
FIG. 29 depicts exemplary embodiments of the digital telephones in further detail, illustrating the physical features that preferably accompany each telephone set. Preferably each telephone consists of telephone chassis 474, handset 476, display 478, feature keys 480, feature key LEDs 482, dial pad 484, speaker 486, volume control keys 488, and microphone jack 490. In preferred embodiments of the digital telephones, the features include tri-colored LEDs, wherein each color corresponds to one of three states (on, off, or flash) and indicates the state of each line key, such as idle, dial, ring, connect, block, hold, bypass, etc. It should be appreciated that in alternate embodiments, the physical features may include additional LEDs, buttons, speakers, microphones, switches, jacks, ports, antennas, card slots, and cardbuses, etc.

Detailed Description Text (239):
The digital telephones preferably provide the following pre-programmed features: (1) Idle: When two telephones are connected, and the user disconnects and returns to idle, the second telephone also returns to idle, so that it is unnecessary for the user to hang up the handset or press release. (2) Hold: To place the current call on hold and answer the call waiting call, the user may press the call waiting button. The primary line LED may flash (e.g.: red) and the call waiting LED may be lit and connected (e.g.: green). To place the call waiting on hold and switch back to the original call, the user presses the primary line key. This feature of toggling back and forth between the two calls can continue as many times as desired. (3) Hot dial pad: While the telephone is idle, the user may press a digit to automatically place the telephone in handsfree mode and initiate a dialing sequence. When the last digit is dialed, then the call is placed. The call may be switched to the handset at any time by lifting the handset. (4) Autodial: Pressing the 1 key while the telephone is idle automatically puts the telephone in handsfree mode and dials the number programmed into the autodial button, thus placing the call. The call progress tones are audible through the speaker. The call may be switched to the handset at any time by lifting the handset. (5) Message waiting: The message waiting indicator doubles as an easily accessible way to call and access voicemail. The user

presses the MWI key while telephone is idle, which automatically places the telephone in the handsfree mode and accesses voicemail. (6) Conference call: For a conference call, the user establishes a connection with the first party, either by receiving or making a call, then presses the conference button and dials the second party. After the call is answered, the user presses the conference button again. The display updates on the telephone on each connected party to show there are three members in the conference. To add a fourth party, the user may repeat these steps, starting with pressing the conference button. (7) Transfer: To transfer a call that is presently connected, the user preferably presses transfer and dial the target extension, waits for the answer, then presses the transfer button again. If the user wishes to perform a blind transfer, then the user presses the transfer button again while the call is still ringing. (8) Blind transfer: If the user frequently does blind transfers, the transfer button may be programmed to always perform a blind transfer. In this case, the second push of the transfer button is unnecessary. (9) Call waiting: For call waiting, a call to an extension while connected in another call will show as ringing on the call waiting button. The call waiting button is then treated as a pseudo-line key.

Detailed Description Text (240):
Referring to FIG. 30, preferred embodiments of the GUI configuration of the programmable digital telephones will now be described. It should be understood that such a configuration application may be run any computer connected to communication system 50, similar to the configuration windows described elsewhere herein. Thus, in addition to programming features and buttons of such digital phones with telephone keypad depressions (with the digital phone put in a program mode that conveys phone configuration information to communication system 50 such as by commands sent via the telephone), such a graphical interface may more desirably guide a user through the steps of programming the digital phones, with the configuration data resulting from the configuration send to communication system 50 such as over a packet bus as previously described.

Detailed Description Text (241):
As illustrated in FIG. 30, configure user window 492 is provided to configure feature keys 480 of digital telephones 466 to 472. Configure user window 492 may be opened by a user or administration of an office attendant-type program or remote configuration-type program running on a computer coupled to communication system 50. Window 492 preferably includes user name display 494, extension display 496, OK button 498, cancel button 500, help button 502, and tabbed windows 504, such as overview, telephone, forwarding, pickup group, etc. To program the digital telephones, the user preferably selects the tabbed window labeled telephone, whereupon telephone window 506 will appear.

Detailed Description Text (242):
FIG. 31 illustrates telephone window 506, which is provided for identifying the features of digital telephones 466 to 472. As noted in telephone window 506, the user is instructed to select the telephone type from a predetermined list of telephone types, which may be limited by the hardware of the slot or port. The list of telephone types appears in pulldown menu 508. The user preferably selects radial button 510 to enable or disable the supported digital telephone. The user may also select the types of features from a predetermined list of templates in pulldown menu 512, which are determined by the type of digital telephone selected by the user. After the user selects a feature from pulldown menu 512, the user may select customize button 514 to modify the programmable fields for the feature key settings on the selected digital telephone. Additional features, such as call waiting and do not disturb, are noted with checkboxes 516, in the features section of telephone window 506. FIG. 32 further illustrates pulldown menu 512, which preferably includes a plurality of features, such as basic, DSS/BLF, retail, secretarial, etc. In accordance with the present invention, data for configuring digital telephones 466-472, which has been selected and/or entered by the user, is saved in communication system 50. Communication system 50 stores configuration data for the particular programmed digital phone, which is then later accessed as the phone is utilized in order for communication system 50 to decode the button depressions on the digital phone and to take appropriate action in response thereto.

Detailed Description Text (243):
With reference to FIG. 33, after customize button 514 is selected by the user, telephone button window 516 appears, displaying a digital image of the selected telephone, such as 16-button telephone 470. (The selected telephone will correspond to the type of telephone chosen by the user in pulldown menu 508 as illustrated in FIG. 31.) Telephone button window 516 provides

programmable fields for each feature key of digital telephones 466-472. By selecting a key in the GUI, the user may choose from a predetermined list of features provided in telephone button window 518. Accordingly, data for configuring digital telephones 466-472, which has been selected and/or entered by the user, is accordingly saved in communication system 50.

Detailed Description Text (244):
As illustrated in FIG. 34, telephone button window 518 preferably includes customizable pop-up menu 520, which indicates the various programmable features available for the digital telephones. Preferably pop-up menu 520 is positioned adjacent to feature keys 480. The programmable features, for example, may include auto dial 522, call return 524, call waiting 526, camp-on 528, Centrex flash 530, Designated station select/busy lamp field 532, do not disturb 534, extension pickup 536, line appearance 538, program 540, self park 542, transfer 544, unassigned 546, user forward 548, and voice call 550.

Detailed Description Text (245):
In accordance with preferred embodiments of the present invention, auto dial 522 is provided to automatically dial a pre-configured telephone number. Call return 524 rings back the extension of the last inbound call, if the call originated within the office attendant-type system. Call waiting 526 places an existing call on hold in order to connect to an incoming call. Camp-on 528 is similar to call return 526, but provides the option of ringing back an extension as soon as the extension becomes available. Centrex flash 530 accesses call transfer features provided by Centrex telephone services, which is available through most TSPs. Designated station select/busy lamp field (or DSS/BLF) 532 monitors the status of specified extensions and transfers calls to those extensions. Do not disturb 534 prevents a telephone from ringing. Extension pickup 536 answers a specific ringing extension within a call pickup group. Line appearance 538 is a secondary line for extensions without a designated station port or a physical telephone; it provides a voicemail rollover function for either primary or virtual extensions. Program 540 enables programmable features, such as auto dial, forward, and voicecall keys. Self park 542 places a call in a parked state on the user telephone for retrieval from another telephone. Transfer 544 transfers calls to another extension. Unassigned 546 provides the option of leaving the extension unassigned. User forward 548 dispatches calls to another extension or telephone number. Voicecall 550 enables an intercom to page a specific extension. It is important to note that some of these programmable features may be selected and used simultaneously, depending upon which features are enabled and disabled.

Detailed Description Text (246):
As further illustrated in FIG. 34, if a feature is selected in telephone button window 518, then the feature preferably is highlighted as exemplified by DSS/BLF 532. The user then preferably presses OK button 552 to make the selection or cancel button 554 to cancel the selection. However, if a feature is selected, then configuration window 556 appears as illustrated in FIG. 35. Accordingly, data for configuring digital telephones 466-472, which has been selected and/or entered by the user, is accordingly saved in communication system 50.

Detailed Description Text (248):
As illustrated in FIG. 36, additional settings window 561 preferably includes one or a plurality of check boxes for optional features, such as do not ring telephone, do not receive paging, use offhook ring, mute microphone when voice calls are received, etc. In accordance with the present invention, data for configuring digital telephones 466-472, which has been selected and/or entered by the user, is accordingly saved in communication system 50.

Detailed Description Text (249):
In addition, similar GUI configuration windows preferably are provided in order to facilitate production of labels that are typically applied to the digital phone in order to provide a visual indication of the particular functions that have been programmed for the particular buttons of the phone, etc. Preferably, an application runs in conjunction with the phone configuration application, which automatically prints labels for the feature keys in accordance with the programmable fields selected for a specified digital telephone. In a preferred embodiment of the present invention, data about programmable fields generated as part of the phone configuration process are stored in one or more files that are accessible by the label generation application. Such files preferably extract the configuration data, associate it with particular buttons of the particular phone that was configured, with the application preparing a label that corresponds to the particular programmed features for the particular programmed phone. Thus, the label generation application may use this data to print programmable field

information (e.g. redial, hold, voice call, etc.) on preferably pre-formatted labels which correspond to the type of digital telephone and the selected feature keys of feature keys 480. Labels may be printed in a plurality of formats of generic telephone labels and in accordance with the programmable features and types of digital telephones. For example, a user may print pre-formatted labels for 16-button telephone 470, so that the pre-formatted labels correspond to feature keys 480 on 16-button telephone 470. Label generation window preferably provides the user with step-by-step instructions for printing labels via a series of linked windows. It is important to note that the label generation window also preferably provides a one-click button for printing labels, so that a button in the label generation window may be pressed by the user to print the feature key label. Thus, a computer, such as computer 24 in FIG. 3, when coupled to communication system 50, may print digital telephone labels via a printer, such as printer 22 in FIG. 3. In accordance with the present invention, data for configuring digital telephones 466-472, which has been selected and/or entered by the user, is accordingly saved by communication system 50.

Detailed Description Text (251):
In accordance with embodiments of the present invention, a variety of highly integrated voice, data, and video communications systems may also be employed. In accordance with additional aspects of the present invention, highly advantageous methods for administering call control and management also are provided, including for Voice over Internet Protocol ("VoIP") type telephone calls.

Detailed Description Text (252):
In accordance with the present invention, an administrator may more optimally control calls made by or to the system in accordance with what herein is referred to as a dialing plan. In particular, the dialing plan in accordance with preferred embodiments of the present invention will provide for improved control over routing of outbound calls. In accordance with preferred embodiments, the dialing plan will provide VoIP call support; global permission and restrictions via a global access profile; area code tables with support for office code ranges; multifunctional route tables, including destination routing, time of day routing; multiple trunk group queuing; trunk overflow; digit Translation; and unified dialing for off-premise extensions (Off-Premise Extension Table).

Detailed Description Text (254):
In preferred embodiments, the access profile (or trunk access profile) may be administered through an administration screen, which preferably is accessed through an Automatic Route Selection ("ARS") screen. In preferred embodiments, access profiles may be edited, created, deleted or copied to other profiles, and extensions are assigned an access profile to specify dialing capabilities for the extensions. Instead of routing to a specific trunk group based on the initial check of the access profile, in accordance with such embodiments system administrators will now be about to direct an outgoing call through various filters. Once the filtering has been completed, the call can be either sent to a single trunk group or sent to a specific routing table with a number of possible steps to be executed.

Detailed Description Text (255):
In such preferred embodiments, there are a number of dialing area tables for dialing area control. In certain embodiments, three dialing area tables are provided, namely an area code table(s), international country code table, and a trunk group access codes table. Such tables are implemented to give the administrator-type person the ability to control where a user may or may not call. In such embodiments, routing tables are used to control access to trunk resources. A call which is being routed via a specific route table prevents the caller from using resources not accessed via the access profile. In preferred embodiments, such routing tables contain the same redundant profile field, but the purpose here is different. In a routing table, a customer uses the trunk access profile to control resources. For example, a user may be allowed to dial a distant area as defined in the area code table, but is restricted to only using the least expensive trunk group in the routing table. Such use of trunk access profiles may be utilized to desirably control access to and use of various telecommunications resources.

Detailed Description Text (256):
Preferably, a screen is provided for "Local Area Codes". Such a local area code table preferably contains: a Home Area Code--7-digit local calls are routed as if this area code was dialed; and a Local Area Code List--Area codes that are treated as local calls. Preferably such

local area codes include area codes not requiring 1+dialing and area codes requiring 1+dialing. Thus, in accordance with the present invention the local area codes table may be accessed so that calls may be more intelligently controlled and managed (e.g., in order to avoid treating non-toll local calls as long distance calls, even though the local calls are dialed with 1+10 digit dialing like traditional long distance calls).

Detailed Description Text (258):
FIG. 37 illustrates Automatic Route Selection ("ARS") window 570 used in preferred embodiments of the present invention. As illustrated, ARS window 570 preferably includes display 572, which displays for the administrator a list of current access profiles in the system. The use of such access profiles is described in greater detail elsewhere herein. ARS window 570 also preferably provides button 574 (selectable with a mouse or pointer, etc.) that may be selected to edit global access profiles. Button 574 is used to add, modify or edit global access settings that preferably are applied to all stations/incoming trunks when an outgoing/tandem (ARS) call is made. As illustrated, window 570 also may include buttons to edit access profiles, add new access profiles, delete access profiles, copy an access profile to another access profile, restore (i.e., ignore changes), apply changes, complete (done) the access to window 570 or to access on-line help information.

Detailed Description Text (259):
Further aspects of global access profiles will now be described. Global access profiles in accordance with the present invention may be considered an enhancement of an emergency trunk access profile. Entries in the global access profile, in effect, override the dialing extension's configured access profile. In preferred embodiment, the global access table contains three tabs (FIGS. 38A-C). The special digits table (FIG. 38A), in preferred embodiments, is the first table checked for outgoing ARS calls. The special digits table allows the administrator to route or block a call based on the initial sequence of digits. The area code table (FIG. 38B), in preferred embodiments, is the second table checked for outgoing ARS calls. The area code table is more specialized than the special digits table (need more description). The off-premise extension table (FIG. 38C) preferably contains routing information for extensions located in another PBX/system connected to the extension (e.g., two such systems connected together, either in the same office or location or in geographically removed offices or locations. The extension range in the off-premise extension table preferably conforms to the first digit table in the particular system. Such tables will now be further described.

Detailed Description Text (261):
An exemplary area code table is shown in FIG. 38B. The area code tables preferably are expandable tables for handling North American Numbering Plan (NANP) numbers. An area code table is preferably accessed via an access profile screen. Every specific profile preferably has an associated area code table. The area code field preferably is a list field. All NANP area codes can be listed here, including an entry for unspecified (default) area codes (i.e., area codes added to the NANP, which were not known when the area code table was created). As for the office code range field, if the dialed number needs to have a more granular examination, then the office code range should be entered in this field. The routing table field specifies the routing table used to send the dialed number out on a trunk. If set to blocked, then the area code is not allowed for the specific profile and the user preferably will hear fast busy.

Detailed Description Text (263):
As previously described, preferred embodiments utilize an access profile to desirably assist in controlling/managing calls in the system. FIGS. 39A-C illustrate windows that preferably are used to manage access profiles. The preferred access profile screen desirably allows the administrator to configure permissions/restrictions for users assigned to the particular access profile. The access profile is used for calls which do not match any of the settings in the global access profile (a match in the global access profile results in the call being routed, intercepted, blocked, etc., as directed by the global access profile table). As illustrated in FIGS. 39A-C, the preferred access profile window contains three tabs: an area code table (all ARS calls preferably are checked in this table); a privileges table (such as international and carrier access calls, which typically include predetermined initial digits for specifying a particular carrier, such 1010xxx-type carrier access calls); and a trunk group access codes table (which preferably is a permission table for using non-ARS trunk group access codes). These access profile windows/tables will now be further described.

Detailed Description Text (264):
FIG. 39A illustrates a preferred area code table for access profiles in accordance with the present invention. The area code tables preferably are expandable tables for handling North American Numbering Plan (NANP) numbers. An area code table is accessed via the access profile screen by selection of the appropriate tab. Every profile has an associated area code table. In preferred embodiments, area code tables are not shared by profiles.

Detailed Description Text (265):
As illustrated in FIG. 39A, an area code field is provided. The area code field is a list field. All NANP area codes may be listed here, and it may include an entry for unspecified (default) area codes (i.e. area codes added to the NANP, which were not known when the area code table was created.) An office code range also is preferably provided. If the dialed number needs to have a more granular examination than just the area code, then an office code or code range may be entered in this field. A routing table also is provided. The route table field preferably specifies the route table that will be used to send the dialed number out on a trunk. If set to blocked, then the area code is not allowed for the profile and the user will hear fast busy (or be provided some other indicia that the call is blocked, such as described elsewhere herein).

Detailed Description Text (270):
Also in accordance with such embodiments, a first digit table is provided to facilitate desirable call control and management in accordance with the present invention. As is known in the art, first digit tables are utilized to process and control calls by way of analyzing first digits dialed, for example, by a user depressing keys on a telephone. In accordance with such embodiments of the present invention, the first digit table is improved in order to more desirably support unified dialing plans and an increased number of trunk access codes as described herein. In addition, since the extension range of the off-premise extension table will follow the first digit table and reflect the extension range of the distant system/PBX, the first digit table preferably supports multiple extension lengths; e.g., first digit 3 may have an extension length of three, while first digit 4 may have an extension length of five, etc. Also in accordance with such embodiments, first digits preferably do not serve to designate as "off-premise."

Detailed Description Text (272):
As illustrated in FIG. 41, the first digit table may include, for example, a first digit tab, which may include first digit/types for attendant, extension, external, or not configured, etc., as illustrated. The attendant first digit/type preferably directs calls to the operator or attendant, whether automated or in person. The extension first digit/type are configured as extensions, which digits collected in accordance with the extension dialing rules, and then appropriately routed. An external digit/type designates a trunk access/NANP digit. Other fields applicable for external calls include the access code, which in preferred embodiments may be a one or two digit access code. The routing field, may include a <Trunk Group> (such as T1, voice digital, voice analog, etc.) designation, where the call will be routed to the specified trunk group. The routing filed may state "not configured," indicating that the access code is not defined (a user dialing this access code preferably will hear fast busy or otherwise be provided with some indication that the access code is not defined, as discussed elsewhere herein). If the trunk field is designated "outbound routing," the call will be routed through an ARS process as discussed elsewhere herein. If the trunk field is designated "outbound routing, the collect digits field is automatically set to NANP. The collect digits field specifies the maximum number of digits to collect after the access code has been entered. If this field is set to NANP, then the North American Numbering Plan is used to determine the number of digits to collect. In preferred embodiments, entering a number in this field allows the caller to indicate he/she has finished dialing using the `#` key or letting interdigit-timing to expire. The dialtone field indicates whether or not dialtone is sent after the access code is received. In preferred embodiments, dialtone will remain until the next digit is received; if expected digits is set to the same length as the length of the access code, dial tone will not be sent.

Detailed Description Text (273):
FIG. 42 illustrates a local area codes screen that may be accessed with a tab from the first digit table window. The local area codes screen preferably is used for setting the home area code and dialable local area codes. In accordance with preferred embodiments, area codes may be listed as "1+Area Code" or "Area Code Only." An "Area Code Only" designation allows a user to

dial the specified area code in a 10-digit number without requiring the user to dial "1" first. A "1+Area Code" causes 11-digit calls with the specified area code to be designated as local calls. In accordance with such embodiments, an administrator periodically checks local requirements to determine which format is appropriate for the particular dialing area. Preferably, both formats are possible within the same system/PBX.

Detailed Description Text (278):
Other aspects of caller ID functionality that are included in certain embodiments of the invention include the following. For automatic call distribution ("ACD") applications, such as a software application that runs on the system (such as described elsewhere herein) that processes calls in a manner to distribute/forward the calls based on caller inputs, such as digits selecting particular departments, caller account number or identifying numbers, etc., voice mail or pager applications, such are preferably mapped to connected to system ports (like a system extension). As system ports, ACD, voice mail, and pager numbers/applications preferably also have caller IDs associated with them. In preferred embodiments, for internal calls, the caller ID for such system ports preferably is the name and number associated with the group in which they belong; for external calls, the caller ID preferably is the company name and number. In addition, function codes may be entered to, for example, activate or block caller ID for particular calls (such as by dialing *67). In preferred embodiments, regardless of settings, calls to 911 or similar emergency type or special calls (e.g., operator) are not blocked, regardless of system settings.

Detailed Description Text (281):
VoIP functionality in accordance with such embodiments preferably provide a predetermined number of IP voice channels, for example up to eight or more channels of IP voice. Preferably, such embodiments provide dynamic support for both H.323 and Media Gateway Control Protocol (MGCP) call-control signaling. Unlike systems with an external gateway between a PBX and a data router, the integrated solution of preferred embodiments of the present invention provide direct conversions between attached telephones and IP trunks without requiring a configuration of several devices. To further enhance voice quality, embodiments of the present invention also preferably include dynamically adjustable jitter buffers, packet-loss correction, and noise-level matching. For greater reliability, traffic can be rerouted over alternative networks if there is a failure to connect over the primary route.

Detailed Description Text (282):
In accordance with such embodiments, and as discussed more fully elsewhere herein, a more consistent user experience is provided with a single, integrated dialing plan for both circuit- and packet-switched voice calls. For example, employees at a branch office may contact co-workers at any other location by simply dialing their extension. The uniform dialing plan simplifies a company's migration to VoIP, since administrators can easily configure calls over any type of connection. With uniform dialing, such embodiments may utilize VoIP calling, when available, in a manner that is transparent to end users. Thus, in accordance with such embodiments, in a headquarters-branch office(s) arrangement, separate access trunks for voice and data do not need to be deployed at each site. Low-cost VoIP calling between, for example, a headquarters site and a branch office, and employees may use the same phones and dialing plans they are accustomed to, and the system automatically converts interoffice calls to VoIP calls if available, and if not can route the calls in a manner transparent to end users through alternative routing. Such systems may support simultaneous interfaces to both packet- and circuit-switched networks for voice calling, and least-cost routing may be automatically enabled for both IP and traditional voice trunks. In accordance with the present invention, using a packet-optimized WAN for telephony transport may significantly reduce costs by sharing expensive WAN bandwidth with data transmissions. Also in accordance with such embodiments, low-bandwidth coder-decoders (CODECs) and silence suppressors can be used to yield, for example, an 8:1 bandwidth savings over standard circuit-switched voice calls.

Detailed Description Text (288):
The H.323 standard, in general, specifies four types of components that may be networked together to provide point-to-point and point-to-multipoint multimedia-communications services--terminals, gateways, gatekeepers and multipoint control units. H.323 terminals generally are used for real-time bi-directional multimedia communications. An H.323 terminal can either be a PC or a standalone device (which could be conference or other telephone or video conferencing unit, such as described elsewhere herein), which is running an H.323 application and the multimedia/communication application(s). In accordance with preferred embodiments, the H.323

terminal provides audio communications while optionally supporting video or data communications. An H.323 gateway provides connectivity between an H.323 network and a non-H.323 network. For example, such a gateway can connect and provide communication between an H.323 terminal and the PSTN. This connectivity of dissimilar networks is achieved by translating protocols for call setup and release, converting media formats between different networks, and transferring information between the networks connected by the gateway. While a gateway is not required, however, for communication between two terminals on an H.323 network, systems in accordance with preferred embodiments provide software/hardware resources to enable the system to serve as an H.323 gateway. A gatekeeper can be considered the brain or intelligence of an H.323 network. Although not required, gatekeepers provide desirable services such as addressing, authorization and authentication of terminals and gateways, bandwidth management, accounting, billing, charging and call routing-type services. Preferred embodiments of the present invention desirably work with an external H.323 gatekeeper, or in alternative embodiments also integrate/provide software/hardware resources to enable the system to serve as an H.323 gatekeeper. Multipoint control units (MCUs) provide support for conferences of three or more H.323 terminals. All terminals participating in the conference preferably establish a connection with the MCU, and the MCU manages conference resources, negotiates between terminals for the purpose of determining the audio or video CODEC that will be used, and may handle the media stream. While gatekeepers, gateways and MCUs typically are considered logically separate components of the H.323 standard, in preferred embodiments various of these components are integrated or tightly coupled to a preferred system as a single physical device.

Detailed Description Text (291):
Using such DSP resources (and other hardware/software resources), an analog voice signal is received or generated by the system (such as by a person speaking into a telephone, which creates an analog voice signal as they speak). The analog voice preferably is converted to a Pulse Code Modulation (PCM) digital stream. (As is known in the art, PCM is a sampling technique for digitizing analog signals, especially audio signals. PCM samples the signal 8000 times a second, and each sample is represented by 8 bits for a total of 64 Kbps. There are two standards for coding the sample level; the Mu-Law standard generally is used in North America and Japan, while the A-Law standard is used in most other countries.) Preferably, hardware/software of the system analyzes the PCM stream, and preferably echo is removed, voice activity detection (VAD) is performed, and tone detection is performed; remaining PCM samples are provided to the codec for processing. The voice codec (which may largely be implemented in software running on a processor) generally is used as part of the process of converting the originally analog signal to digital data packets suitable for transmission over a data network. In accordance with preferred embodiments of the present invention, different software codecs may be used in the process to convert analog signals to digital data in frames (also providing various levels of data compression; e.g., a G.729a frame may be 10 ms long and contain 10 bytes of speech), and then convert digital data back to analog signals. The goal when selecting a codec is to maintain high voice quality and low latency. Generally speaking, lower bit rate codecs offer higher complexity and, therefore, introduce higher latency. As a result, tradeoffs are made between the goals of low bit rate, high quality, low complexity and low latency. The actual choice will depend on the particular application and quality and bandwidth concerns. The various codec standards (i.e., the G. standards discussed earlier, publicly available standards data for which is hereby incorporated by reference, as with the other standards-related information for standards referenced herein), generally may be evaluated and selected on the basis of criteria such as bit rate, quality, complexity, bandwidth usage or frame size and latency. In preferred embodiments, a G.723.1 codec often is utilized. Packet assembler software preferably running on one of the DSP (within the provided DSP resources) receives frames from the codec and creates packets. Several frames may be combined in a single packet. In preferred embodiments, a 12 byte RTP header is added to each packet, which provides a sequence number and time stamp, and the packet thereafter is forwarded, preferably to a host or other processor for further processing.

Detailed Description Text (292):
Addressing in VoIP is provided in a manner to determine from the dialed digits, preferably identified in the DSP, the destination IP address (e.g., 301-236-1895.fwdarw.193.148.100.2). Such as from a lookup table under control of a processor (as part of the software/hardware resources of the system), a preferably 20 byte IP header is added to the packet, which contains (1) the IP address of the source system/gateway, and (2) the IP address of the destination system/gateway. An 8 byte UDP header containing source and destination sockets also is added. Systems (such as described herein or otherwise) on the network may then examine the IP address

to identify the route to the destination. It should be noted that several systems as described in the preferred embodiments herein may be in the path that the packets take to their destination.

Detailed Description Text (293):
Among the problems encountered in VoIP communications are delay and echo. Delay causes problems such as echo and talker overlap. This problem is illustrated in FIG. 44. Echo typically is caused by signal reflections of the speaker's voice at the far end telephone equipment back into the speaker's ear. This echo is caused by a device called a hybrid, which typically is a 4 wire to 2 wire converter. The telephone handset has 2 wires going to the ear piece and 2 wires going to the mouth piece, and inside the telephone those 4 wires need to be converted to only 2 wires, which is what the telephone network typically uses (i.e., tip and ring). Echo generally becomes a significant problem when the round trip delay becomes greater than 50 milliseconds. Since echo is perceived as a significant quality problem, voice over data systems desirably will address the need for echo control and implement echo cancellation. Talker overlap, or the problem of one talker "stepping" on the other talker's speech, becomes significant if the one-way delay becomes greater than 250 ms. The end-to-end delay budget is therefore a significant consideration, and delay must be reduced through a packet network, which the present invention attempts to achieve.

Detailed Description Text (297):
As an example, if a particular system/gateway negotiates a Rx packet time of 10 mSec with a remote system/gateway, in preferred embodiments the jitter buffer is automatically set to be in effect 30 mSec. Thus, without requiring an administrator to optimally set the jitter buffer size on a call-by-call basis, and without setting the jitter buffer size (or range) to be undesirably large to accommodate the largest possible desired size, in preferred embodiments the jitter buffer size/range is set automatically upon negotiation of the codec parameters, etc. Thus, codecs may be changed (such as automatically and on a preferably call-by-call or destination by destination basis), and a more optimal jitter buffer is automatically determined. More generally, as a VoIP call is set up (again, preferably on a call-by-call basis), which includes multiple protocols such as H.323 or MGCP, the particular codecs and codec parameters (which may be set by preferences on a destination by destination basis), a more optimal jitter buffer setting is automatically determined without further administrator intervention. Thus, protocol and code preferences may be determined such as by particular location/destination, and as calls are initiated by ordinary users, VoIP parameters are automatically retrieved (such as prioritized codec settings) and determined (such an automatically sized jitter buffer).

Detailed Description Text (299):
In addition, as congestion in the network may cause some packets to be dropped, left untreated the listener hears undesirable pops and clicks, etc. This is because IP networks do not guarantee service, lost packets can frequently occur. Under peak loads and congestion in the network, voice frames may be dropped equally with data frames (data frames, however, are not time sensitive and dropped packets may be corrected through the process of retransmission, etc.). In accordance with the present invention, to compensate for lost packets the system preferably replays the last packet received during the interval when the lost packet was supposed to be played out, which in a relatively simple manner fills the time between non-contiguous speech frames. Desirably, the DSP in preferred embodiments plays the last successfully received packet at a decreased volume (e.g., around a 3 dB reduction, or 1/2 volume) in order to fill the gap. For multiple lost packets, the previously received packet may be replayed over and over at a more decreased volume to silence, which has been determined to be much more desirable than a sudden drop. Out of order packets also may result, given that the packets may take diverse routes through a network and may arrive out of order. In accordance with the present invention, out of order packets are not played in the order that they are received; if an out of order packet condition is detected, the missing packet is replaced, preferably, but its predecessor as if it is lost. When the late packet finally arrives, it generally is discarded. With such replacement algorithms for lost/out of sequence packets, and with automatic sizing of the jitter buffer, more optimum VoIP communications may be produced.

Detailed Description Text (307):
One way to deal with this problem is to have one set of routing settings for all IP Telephony calls no matter where the call is going or coming from. This option is very limiting and undesirable. The other option is to define a way to specify source and destination information

for IP telephony calls. After defining source and destination, routing parameters may be associated with those defined sources and destinations. In other words, logical addressing of IP telephony calls will define the source and destination of the call and not the physical trunk that carries the voice data. This logical addressing is obviously the IP address or anything that resolve into the IP address of the source or destination of the IP telephony call.

Detailed Description Text (308):
An IP Call Destination contains the IP address or a range of IP addresses of the end point's IP telephony gateway. It also contains all the IP telephony call specific settings that the IP telephony manager software needs to use in order to make a successful call to that destination gateway, for example, codec preference order. This destination can be used instead of an outbound Voice Trunk Group anywhere a Trunk Group is used in the outbound routing algorithm settings. Specifically, in First Digit Table an IP Call Destination can be selected instead of a Trunk Group for the call to be routed to. Also IP Call Destination can be selected instead of an outbound Voice Trunk Group in outbound routing table steps.

Detailed Description Text (309):
An IP Telephony Source contains the IP address or a range of IP addresses of the source gateway (where the call is coming from). It also contains the settings that inbound routing algorithm needs to know to successfully route the call. These settings are similar to information that is stored in an inbound Voice Trunk Group for a traditional call. To make it easier for the administrator of the system to reuse these settings, the IP telephony Source configuration section preferably is divided into IP Source settings and Source IP Address sections. This way the administrator can have reusable set of settings that can be associated with several source IP addresses. Preferably, in accordance with the present invention this is similar to what Trunk Groups do for trunks. One of the main parameters in these settings is the Inbound Routing Table which is exactly the same as defined for traditional telephony.

Detailed Description Text (312):
Inbound call routing for IP Telephony calls preferably uses the IP address of the source gateway to determine which IP Source Settings to use. This is very similar to a traditional call in which case the trunk that the call is coming from determines which inbound Voice Trunk Group to be used for inbound call routing. After this step everything is the same for IP Telephony inbound call routing and the traditional inbound call routing.

Detailed Description Text (314):
A database table preferably is provided for the IP Call Destination. Each record in this table preferably defines a destination. Each IP Call Destination preferably has an ordered list of codec preferences, and preferably includes fields such as destination ID and/name, destination IP address, caller ID format, protocol (e.g., H.323 or MGCP), jitter buffer size parameters (see also the discussion elsewhere herein regarding jitter buffer sizing), send/receive volume settings, echo cancellation settings (e.g., filter size), silence suppression settings, voice activity detection settings (e.g., threshold) and the like.

Detailed Description Text (315):
As it was mentioned before, there are two main paths that are preferably used to configure an outbound call to go over IP telephony. These two paths are: through an outbound routing table step or through First Digit Table. The preferably graphical interface for outbound Routing Table configuration preferably will merge the list of available IP destinations with the list of outbound Trunk Groups and present the combination to an administrator when they are selecting the destination for a routing table step. IP destinations, if chosen, will be used by the routing algorithm to route the call over the IP telephony. This is an important aspect of the present invention. The routing table may treat the VoIP telephony route as a step in the routing table. Thus, and as previously described, an assessment may be made of whether suitable conditions exist for a VoIP type call to be made, which may be made by attempting to establish a VoIP connection via a first step in a routing table. If the VoIP fails to complete, the system may automatically go to a next step in the routing table, which could be another VoIP step but more typically would be a type of traditional telephony Trunk Group/destination. Thus, in a desirable and automated manner, a VoIP call may be attempted, with a first step in the routing table, and with a traditional telephony call attempted (i.e., go the next step in the routing table) in the event of failure of the VoIP call, etc. This is illustrated in FIG. 46C, which shows a partial routing table with an initial IP telephony step, followed by voice

digital and voice analog Trunk Group steps, etc. The First Digit Table user interface preferably will merge the list of available outbound Trunk Groups with the list of available IP destinations and present it to user as the list of destinations. IP destinations, if chosen, will be used by the routing algorithm to route the call over the IP telephony.

Detailed Description Text (316):
IP telephony inbound calls will be tagged by the IP address of the source of the call. Other than this, they preferably have all the characteristics of a non-IP call. Inbound routing algorithm determines if the call is an IP call based on the IP tag in which case it tries to match the source IP with one of the IP address patterns stored in a Source IP Address table. Based on this match it can determine the IP Source Settings record associated with this IP address. From this point on, an IP Source Settings record can be used like an inbound Trunk Group and inbound routing algorithm continues to route the call same way as a non-IP call. There are two tables preferably associated with inbound IP Telephony call configuration. Source IP Address table and IP Source Settings table. The Source IP Address Table preferably stores all the settings associated with a source IP address. The IP Source Settings Table preferably stores all of the information that inbound routing algorithm needs to know in order to be able to route the call correctly.

Detailed Description Text (319):
FIG. 47A illustrates two communication systems 50 in geographically remote locations, such as in San Francisco and New York. In a first scenario (indicted by the solid line), a call may be made from an extension of the San Francisco communication system 50 to an extension of the New York communication system 50. Note that the numbers dialed may be a simple extension as is to a local extension. Based on the call routing and IP telephony parameter selections as described earlier, an IP telephony call may be established with the New York extension, which may be completed with the user simply dialing a simple (e.g., four digit) extension number. In a second scenario (indicated by the dotted line), a call may be made from an extension of the San Francisco communication system 50 to and an external telephone reached via a PSTN through the New York communication system 50. As illustrated, the user may dial a number for a phone in New York, and the San Francisco communication system 50 routes the call as an IP telephony call to the New York communication system 50, which then completes the call over the PSTN. This call may offer cost advantages in that it avoids the long distance toll charges in going from San Francisco to New York. In a third scenario (indicated by the dashed line), a call is made from an external telephone coupled to the San Francisco communication system 50 via the PSTN to an extension of the New York communication system 50. Note that the user of the external extension dials a local number for the San Francisco communication system 50, which may then route the call as an IP telephony call to the New York communication system 50.

Detailed Description Text (321):
Referring now to FIG. 47B, an additional scenario in accordance with preferred embodiments of the present invention will now be described. In this exemplary scenario, an H.323 gatekeeper is coupled to the IP network. As an initial step of the scenario, a user enters an extension on a telephone coupled to the San Francisco communication system 50, which knows that the destination for this extension is an extension of the New York communication system 50. Based on the routing table information, etc., the San Francisco communication system 50 may then request a telephone number to IP address resolution from the H.323 gatekeeper coupled to the IP network. The call may then be routed by the San Francisco communication system 50 to the New York communication system 50 based on the IP address received from the H.323 gatekeeper. In such a scenario, various desirable aspects of preferred embodiments such as routing table selection, codec and codec parameter selection, jitter buffer sizing and the like (as described earlier herein) may be utilized in order to provide more optimal and cost effective IP telephony solutions.

Detailed Description Text (327):
In alternate embodiments, a mechanism is provided to prevent undesirable affects that could result if both sites attempt to switch over at the same time. In one such embodiment, assuming that both locations are equally capable of detecting network problems, the switchover is always initiated by the side with the "smaller" IP address. In other embodiments, as the IP call is set up, it is determined in advance which system shall be responsible for detecting IP network problems and initiating the switch over.

Detailed Description Text (331):

As illustrated in FIG. 49C, however, in alternative embodiments an address of a particular card is read, where the card has been designed to have one address that simply allows the data bus pull up/pull down information to flow through to the requesting unit. For example, a processor on motherboard 624 could attempt to read a particular address of exemplary resource switch card 623. The particular address of resource switch card 623, however, is designed so that the various elements of resource switch card 623 that may couple to either of buses 626 or 620 do not respond, while bus interfaces 628 allow the backplane encoded pull up/pull down information to pass from data bus 620 through resource switch card 623 to bus 626 (which may be a standard PC-type bus, such as an ISA bus) and motherboard 624. Thus, motherboard 624, which may be accessed remotely (as described elsewhere herein), may "poll" the hardware and determine the backplane encoded information by reading the particular address (the use of such a particular address is illustrated in FIG. 49D, which illustrates a bus read address of the location of the memory map allocated to resource switch card 623).

Detailed Description Text (337):
An exemplary process flow for use of such a deployment disk is as follows. The system may be shutdown, and the system restarted with the deployment disk inserted into the CD ROM drive while the system is booting up (in certain embodiments, certain cards may be removed from the system that are not necessary for software deployment). Upon bootup, the system preferably provides a beep or series of beeps that provides an audible indication that the deployment disk is in the CD ROM drive and has been recognized by the system. Preferably, one or more cards in the system, such as a resource switch card, will provide visual feedback, such as by blinking LEDs, which indicate that the system is ready to proceed. Preferably, a physical button must be depressed for a sufficient, predetermined length of time, which serves to ensure that the installer intends to deploy the software. Preferably a change in the audible output occurs so that feedback is provided, which may be accompanied by other visual feedback, that the deployment process has properly commenced. Initially, the program preferably verifies the system BIOS, and the model key, and upgrades them if necessary (this may be accompanied by a check of backplane encoding information as well). Thereafter, files from the CD ROM preferably are copied onto a hard disk of the system. If successfully deployed, an audible tone is emitted to indicate successful deployment, which is preferably accompanied by visual feedback as well. Still preferably, the CD ROM is automatically ejected at the end of the process (typically the system must be rebooted in order to operate in accordance with the newly deployed software).

Detailed Description Paragraph Table (1):
TABLE 1 Label Where Description InCalls TUI Number of incoming calls answered (all types) MsgCreate MSS Number of messages created MsgSent IMDA Number of messages sent successfully MsgSendFail IMFSA Number of message send failures caused by an error in the Msg Subsystem MsgDelete MSS Number of message deleted MbxLogon MSS Number of times users logged on successfully MbxLogoff MSS Number of times users logged off their mailbox (versus abandoned) TooManyErrors TUI Number of times callers were dropped because they made too many errors TooShort TUI Number of times messages recorded were too short Restart TUI Number of times the AA/VMS application was restarted/reloaded MWIOn MSS Number of requests to turn MWI On MWIOff MSS Number of requests to turn MWI Off MWIFail MSS Number of MWI (On/Off) requests that failed TMOOper TUI Number of calls transferred to Operator because of TMO ZeroOper TUI Number of calls transferred to Operator because caller dialed "0" ErrorOper TUI Number of calls transferred to Operator because of too many errors ErrorPassword TUI Number of calls dropped because of to many password errors. DiskFull MSS Number of times disk was too full to take a message ExtDirInCall TUI Number of direct external (trunk) calls into AA/VMS ExtFwdInCall TUI Number of external calls forwarding into AA/VMS IntDirInCall TUI Number of direct internal (station) calls into AA/VMS IntFwdInCall TUI Number of internal calls forwarding into AA/VMS NewMsg TUI Number of "new" messages recorded and sent by logged on users FwdMsg TUI Number of "forwarded" messages recorded and sent by logged on users ReplyMsg TUI Number of "reply" messages recorded and sent by logged on users MultAddress TUI Number of messages sent that had more than one address NameRecord TUI Number of times a Name message was recorded GreetRecord TUI Number of times a Greeting message was recorded

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L6: Entry 1 of 1 File: USPT Apr 6, 2004

DOCUMENT-IDENTIFIER: US 6718177 B1
TITLE: System for communicating messages via a forward overhead control channel for a programmable logic control device

Brief Summary Text (8):
In view of the foregoing, there is a need for adapting the paging mechanism of a CMR system to support the transfer of commands and/or data for communications from an MSC to a cellular-compatible device. There is a further need for communicating with and controlling the operations of a remote controller, such as a PLC, coupled to a cellular-compatible device that can accept commands and/or data via the paging mechanism of a CMR system.

Brief Summary Text (12):
In response to receiving at least one data page having a predetermined characteristic during the second time period, the cellular communications device can collect the data carried by the received data page. This data can be stored at the cellular communications device for subsequent use or is forwarded for processing by another device, such as a programmable logic controller (PLC). To acknowledge receipt of the data, the cellular communications device can send an acknowledgement signal via the cellular network control channel. This acknowledgement signal can be formatted as a cellular telephony Autonomous Registration signal and carries an indication of data verification. A mobile switching center (MSC) typically receives the acknowledgement signal via the Reverse Overhead Control Channel (RECC) of the cellular network control channel.

Detailed Description Text (10):
It is known that when a cellular mobile radiotelephone originates a call, it transmits a series of data messages to the serving cell. These messages, commonly referred to as Call Origination, are defined by EIA/TIA-553. These data messages contain the low order seven digits of the unit's telephone number, known as the Mobile Identification Number (MIN), the unit's Station Class Mark (SCM), which identifies functional characteristics of the unit, and the Called Address, or dialed telephone number. Cellular system operators typically also require additional data words to be transmitted that contain the MIN2, which is the high order three digits or NPA of the cellular unit's telephone number, and the Electronic Serial Number (ESN). The MIN is assigned to a particular radiotelephone unit by the cellular service provider selected by the subscriber. The MIN typically contains information unique to the CMR system operator, for example, the first three digits of the MIN ("XXX") typically correspond to an area code, the next three digits ("XXX") typically correspond to a geographic location within the area code; and the final four digits ("XXXX") identify a particular piece of equipment. Similarly, the ESN is unique to each mobile cellular radiotelephone unit, and comprises a format that allows differentiation as to manufacturer and, in some cases, the model number, date of manufacture, and the like.

Detailed Description Text (11):
These messages are provided first to the cell, and then through a data link to a mobile telephone switching center, otherwise described as a mobile switching center. The MSC, also known as a MTSO or a "switch," makes voice connections between mobile radiotelephones and other telecommunications networks. At the MSC, a determination is typically made whether the radiotelephone is an authorized user or subscriber by looking up the unit's telephone number, serial number, and other information supplied by the radiotelephone to see if there is an entry in the MSC's database corresponding to that particular telephone. An optional function of an MSC is to validate that the ESN and MIN received as part of a Call Origination message are valid. If the MIN is valid and the radiotelephone is identified as a subscriber within the given cellular system, i.e., a "home" unit, the received ESN is compared to the MSC's database

ESN entry to detect fraud. If these checks succeed, the cellular call is then allowed to proceed.

Detailed Description Text (12):
It is also well known that when a mobile radiotelephone first powers up or first enters a CMR system when already powered, the unit can identify itself as actively present within the system. The radiotelephone identifies itself or "registers" through a process known as Autonomous Registration by supplying a data packet similar to that of a Call Origination message. The Autonomous Registration signal, also referred to as a registration or identification signal, typically comprises data fields for at least a mobile telephone number, i.e., the MIN, and an ESN. The original design attempt of Autonomous Registration was to improve the efficiency of potential future call deliveries by keeping the MSC informed of the approximate whereabouts of each individual radiotelephone unit, and to reduce paging channel load by lessening the need to page all cells to find a particular cellular unit. When the MSC is thus informed, it can later "page" or attempt to ring the cellular unit only in the cell or area that it was last known to be in. Additional cells would be paged only if the initial page did not locate the particular radiotelephone. Thus, Autonomous Registration is simply a set of messages periodically and autonomously sent from the mobile radiotelephone to the serving cell at an interval specified in data parameters previously received from the cell by the cellular unit.

Detailed Description Text (17):
The data message system 10 includes a set of data reporting devices 29, each comprising at least one controller 32' and a cellular communications device 34. A cellular communications device 34 can communicate with the MSC 24 via a control channel of the CMR system. The controller 32', which is connected to one or more instruments or controllable items via a signal path 31', is typically implemented as a PLC for controlling the operations of a field instrument or a controllable item. The cellular communications device 34, which is connected to the controller 32' via a signal path 33', can communicate with the MSC 24 via a cellular network control channel 38 and accept page messages containing commands and/or data for use by the controller 32'. In addition, the cellular communications device 34 can transmit data messages, typically formatted as a Call Origination message or an Autonomous Registration signal, to the MSC 24 via the cellular network control channel 38.

Detailed Description Text (36):
By using the data message format associated with an Autonomous Registration signal, the cellular communications device 34 "registers" with the MSC 24 by sending a data message that appears to contain a mobile telephone number and an ESN. Although it is not intended for the cellular communications device 34 to place a conventional voiced-based cellular telephone call, the cellular communications device 34 nevertheless registers for operation with the MSC 24, thereby enabling the communication of the selected data from the field.

Detailed Description Text (37):
Alternatively, the format for the data message can be identical to the format or data record for a Call Origination signal that is transmitted by a cellular radiotelephone when it originates a telephone call. Similar to the format for a registration signal, the cellular communications device 34 can appear to originate a call by sending a data message formatted as a Call Origination signal to the MSC 24. Although the MSC 24 processes the data message as if it contained a mobile telephone number and an ESN, the data message is actually used to communicate selected data placed within one or more data fills normally reserved for the mobile telephone number and the ESN. Although the Call Origination signal format can be used to transport data from the cellular communications device to the MSC, it will be understood that the data message system 10 is employing this format for data communication rather than for call origination.

Detailed Description Text (38):
As shown in the data record 50 in FIG. 2, the standard message format for a registration signal (call origination) has been adapted by the data message to permit the identification of the particular transmitting cellular communications device 34 and the communication of the selected data. In particular, the data field 52 for the predetermined identifying characteristic corresponds to at least a portion of a mobile telephone number or MIN assigned to the cellular communications device 34. Thus, the predetermined identifying characteristic is substituted within the data field normally reserved for the MIN in an identification signal. This

predetermined identifying characteristic can belong to a set of unassigned mobile telephone numbers. Alternatively, the predetermined identifying characteristic assigned to each cellular communications device 34 can be a conventional telephone number or a set of 10 digits. The predetermined identifying characteristic permits the identification of the source of the data by uniquely identifying the cellular communications device 34. The predetermined identifying characteristic also supplies information used by the MSC 24 to recognize that the data message containing this predetermined identifying characteristic is associated with the data collection system 40.

Detailed Description Text (54):
The receiver 78 can receive pages from the MSC 24 via the FOCC of the control channel 38. For example, the MSC 24 can output command signals, which are formatted as pages, via the control channel 38 to initiate certain operations or to control certain functions of one or more of the cellular communications devices 34 within the cell 12. The receiver 78 can monitor the control channel 38 for finite time periods defined by a clock signal output by the clock 82. For example, the clock 82 can operate as a timer defining a time period for a monitoring operation completed by the receiver 78. The cellular communications device 34 can respond to a command signal by conducting a particular operation or by controlling a certain function associated with the command signal.

Detailed Description Text (56):
The command signal is preferably a 10 digit number that represents a conventional mobile telephone number. At least a portion of this telephone number can be assigned as an identifier for a corresponding cellular communications device 34. The remaining portion (if any) of the 10-digit telephone number can represent a command or data for a particular operation or function. In this manner, a cellular communications device 34 can be programmed to respond only to a command signal containing its address data and to conduct the particular operation or function identified by the command.

Detailed Description Text (59):
The opportunity for placing a command or data, or a combination of a command and data, within the conventional format of a paging message is limited by the defined character length of the paging message, typically the 10-digit telephone number or MIN. For relatively short data lengths, the transmission of a single independent page message from an MSC to a cellular device in the manner known to the art is useful to support limited communications. This fixed data length for a page message is satisfactory for the paging communication task of conventional CMR system operations, namely, the polling of one or more mobile radiotelephones within the coverage area of the CMR system. This polling technique only requires the transmission of a single discrete page message to prompt a response from a mobile radiotelephone unit that receives the page. Prior to the present invention, there was no readily available mechanism for exploiting the paging message mechanism to transfer an expanded data set.

Detailed Description Text (103):
If a programming change occurs locally, typically by the end user or a field technician, in a schedule look-up table, then the target radio can issue a registration sequence to the MSC. In turn, the MSC can forward this information to the data collection system as an update for the schedule information maintained at an Operations Center responsible for schedules.

Detailed Description Text (107):
In decision step 735, a determination is made whether the cellular communications device has received additional scheduled command pages containing data during a predetermined time period. If the response to this inquiry is negative, the "NO" branch is followed from step 735 to step 710 to continue monitoring operations at the cellular communications device. If, on the other hand, the cellular communications device has received additional schedule command pages in a timely fashion, the "YES" branch is followed to step 740. For an exemplary embodiment, programming data is typically carried by this pair of additional scheduled command pages. The programming data carried by the additional scheduled command pages is extracted by the cellular communications device and forwarded to the PLC in step 740. This programming data can be combined with the slot number and programming data received by the PLC in step 725 to support a reprogramming operation at the PLC. For example, the combination of programming data can be stored by a look-up table at the slot number identified by a received scheduled command page to update or to reprogram the programming data at that look-up table slot.

L9: Entry 1 of 5 File: USPT May 10, 2005

DOCUMENT-IDENTIFIER: US 6891940 B1
TITLE: System and method for providing remote access to telecommunications services

Abstract Text (1):
A system and method are provided for reviewing and updating a subscriber's telecommunications services, including a Caller ID service, using a graphical user interface via multiple data networks. The method includes presenting service data to the subscriber via the data networks and transmitting a data message from the subscriber to an intelligent peripheral via at least one of the data networks. The data message indicates a subscriber's desired update to a selected telecommunications service. The method also includes converting the data message into a protocol compatible with an integrated service control point. The converted data message is identical to a data message that the intelligent peripheral would create if the subscriber had entered the desired update via an interactive voice response system. The method further includes transmitting the converted data message to the integrated service control point, and updating the selected telecommunications service in accordance with the subscriber's desired update. Thus, the selected telecommunications service is updated substantially contemporaneously with the subscriber requesting the update at the graphical user interface. Moreover, the subscriber retains the ability to update and review service data via an interactive voice response. The method and system also enable the subscriber to view Caller ID information while being located remotely from the destination of the telephone call associated with the caller ID information.

Brief Summary Text (3):
The present invention relates to the field of telecommunications. More particularly, the present invention relates to a Personal Call Manager, a.k.a. Personal Communications Manager (PCM) providing subscribers integrated access to communications services through a data network, such services include a Remote Access to Caller Identification (RACLID) system. The RACLID system enables subscribers to review caller identification information associated with incoming calls to the subscriber's telephone line from a remote location.

Brief Summary Text (5):
The written description provided herein contains acronyms which refer to various telecommunications services, components and techniques, as well as features relating to the present invention. Although some of these acronyms are known, use of these acronyms is not strictly standardized in the art. For purposes of the written description herein, the acronyms are defined as follows: Advanced Intelligent Network (AIN) Authentication/Subscription Information (ASI) Caller Identification (Caller ID) Customer Premises Equipment (CPE) Dual Tone Multi-Frequency (DTMF) Graphical User Interface (GUI) Generic Data Interface (GDI) HyperText Mark-Up Language (HTML) HyperText Transfer Language Protocol (HTTP) Incoming Call Manager (ICM) Integrated Service Control Point (ISCP) Interactive Voice Response (IVR) Java Database Conductivity (JDBC) Lightweight Directory Access Protocol (LDAP) Line Information Database (LIDB) Outgoing Call Control (OCC) Personal Computer (PC) Personal Call Manager/Personal Communications Manager (PCM) Personal Identification Number (PIN) Public Switched Telephone Network (PSTN) Remote Access to Caller Identification (RACLID) Service Management System (SMS) Service Node (SN) Service Switching Point (SSP) Signaling System 7 (SS7) Signaling Transfer Point (STP) Terminating Attempt Trigger (TAT) Transaction Capabilities Application Part (TCAP) Transmission Control Protocol/internet Protocol (TCP/IP) User Interface (UI) World Wide Web (WWW)

Brief Summary Text (7):
Currently, subscribers to call control services within the Public Switched Telephone Network (PSTN) are able to initiate and modify their services by calling a customer service representative or by interacting with an Interactive Voice Response (IVR) system using a

standard Dual Tone Multi-Frequency (DTMF) telephone device. These methods practically limit the number and types of services that can be provided to and modified by the subscribers because all information pertaining to the services is presented audibly. In addition, the potential market for subscribers to call control services is not fully exploited because of customer reluctance to use IVR systems. An additional drawback is that, conventionally, each PSTN service has a corresponding IVR interface, so that as a customer subscribes to additional services, he or she must keep track of additional IVR telephone numbers and Personal Identification Numbers (PINs).

Brief Summary Text (8):
There have been attempts to remedy the problems associated with IVR access to PSTN services. These attempts incorporate use of packet switched data networks, such as the Internet, to avoid conventional IVR systems and to streamline the initiation and modification functions. The current Internet based systems have several drawbacks, however, including the inability to ensure near real-time update of services and incompatibility with existing IVR implementations.

Brief Summary Text (9):
For many call control services, the subscribers must submit requests to the customer service arm of their provider to initiate new services or update existing ones. The requests are implemented according to the provider's time line and discretion. It is difficult for the users to gauge when the service alteration will take effect. Also, because the current Internet based systems operate exclusively from the conventional IVR systems, i.e., the two systems cannot coexist, customers must select either the Internet interface or the IVR interface. Consequently, a customer who has selected the Internet interface, and who is without a PC and/or Internet access, is not able to make desired changes to his or her services through an IVR. The inability to implement desired changes is especially troublesome considering that users are often interested in altering some call services (e.g., call forwarding, paging, and caller ID) when they are away from their home or business telephone and PC.

Brief Summary Text (10):
An example of call control services provided over a packet switched data network is described in CHANG et al., U.S. Pat. No. 5,958,016, which teaches enabling Advanced Intelligence Network (AIN) services over the World Wide Web (WWW) through a provisioning system called the Service Management System (SMS). The SMS as disclosed in CHANG et al., however, does not ensure near real-time data update and is not compatible with existing IVR implementations.

Brief Summary Text (11):
Therefore, the services presented via the Web are limited in functionality to the extent near real-time data updates are not guaranteed. For example, if a subscriber modifies an incoming call service, which blocks calls from selected phone numbers or classes of phone numbers, to add an allowed incoming phone number, the subscriber will not begin immediately to receive calls from the previously blocked phone number. Rather, the subscriber must wait an unspecified period of time for the service to be updated via the SMS. Also, as discussed above, the Web interface and the IVR interface are mutually exclusive.

Brief Summary Text (12):
The present invention pertains to a Personal Call Manager, a.k.a. a Personal Communications Manager (PCM) system that resolves these problems, simply and efficiently. The PCM provides an interface to telecommunication services, such as personal directories, Incoming Call Manager (ICM), Outgoing Call Control (OCC) and the like. In addition, the PCM interfaces to an improved caller identification (Caller ID) system, referred to as Remote Access to Caller Identification (RACLID). Conventional Caller ID services provided through the PSTN necessitate the attachment of Customer Premises Equipment (CPE) to a telephone jack corresponding to the telephone number (s) subscribing to the Caller ID service. The user may review a log of Caller ID information associated with incoming calls by physically reviewing the information displayed on the CPE. Typically, the Caller ID information includes the name and/or number of the calling party, as well as the date and time of the incoming telephone call.

Brief Summary Text (13):
A limitation of the conventional service is that, in order to review the Caller ID information, the subscriber must be present at the CPE. It would be advantageous, however, for subscribers to be able to review their Caller ID information remotely, e.g., at work, while commuting, on

vacation, etc. Because callers do not always leave messages on an answering device or service, which may be remotely accessible, a subscriber cannot determine through the conventional Caller ID service who has attempted to call until the subscriber physically returns and views the CPE. Consequently, the conventional Caller ID system has several drawbacks, including delayed awareness of incoming telephone calls and subsequently delayed response to those calls.

Brief Summary Text (15):
However, these attempts have several inherent disadvantages. For instance, in both DANNE et al. and VOIT et al., processing of the telephone call is interrupted in order to perform the Caller ID function. Also, the methods provide Caller ID information only when the call is in progress, and in the case of DANNE et al., only when the user is online and running a JAVA application. That is, the user cannot obtain the Caller ID information at his or her convenience. Finally, a significant portion of the intelligence aspects of the DANNE et al. Caller ID system is required to be in the terminals, thus limiting the types of devices that can access the Caller ID information.

Detailed Description Text (3):
An aspect of the present invention provides a user/subscriber access to a PCM system through a communications network, including the Internet and other data networks, without excluding the possibility of conventional IVR access. Thus, the subscriber can conveniently customize services managed by the PCM through a graphical user interface (GUI) that efficiently presents the complex data associated with the managed services with minimal service provider interaction. Another aspect of the invention provides for updating the actual service data in the PSTN substantially contemporaneously with access to the service data via the PCM, permitting near real-time access to the services managed by the PCM.

Detailed Description Text (4):
In another aspect of the present invention, the PCM manages multiple services, including, for example, Caller ID. Thus, the present invention provides the subscriber access to Caller ID information remotely over the communications network in an efficient and user-friendly manner.

Detailed Description Text (5):
According to another aspect of the present invention, a method is provided for reviewing service data relating to a subscriber's telecommunications services using a graphical user interface. The method includes transmitting a data message from the subscriber to an intelligent peripheral through at least one data network, the data message indicating a subscriber's desire to review the service data, and converting the data message into a protocol compatible with an integrated service control point. The converted data message is identical to a data message that the intelligent peripheral would create if the subscriber had indicated the desire to review the service data via an interactive voice response system. The protocol may be the SR-3511 protocol. Then, the converted data message is transmitted to and the service data is retrieved from the integrated service control point. The service data is forwarded to the subscriber through the intelligent peripheral. The subscriber retains the ability to review service data through an interactive voice response.

Detailed Description Text (6):
In another aspect of the present invention, a method is provided for reviewing and updating a subscriber's telecommunications services using a graphical user interface through multiple data networks, including presenting service data to the subscriber through the data networks and transmitting a data message from the subscriber to an intelligent peripheral through at least one of the data networks. The data message indicates the subscriber's desired update to a selected telecommunications service. The data message is converted into a protocol compatible with an integrated service control point, which protocol includes the SR-3511 protocol. The converted data message is identical to a data message that the intelligent peripheral would create if the subscriber had entered the desired update through an interactive voice response system. Then, the converted data message is transmitted to the integrated service control point and the selected telecommunications service is updated in accordance with the subscriber's desired update. The selected telecommunications service is updated substantially contemporaneously with the subscriber requesting the update at the graphical user interface. Also, the subscriber retains the ability to update and review service data through an interactive voice response. The presentation of service data may include retrieving the service data from a service status database, which is periodically updated by the integrated service control point. This reduces traffic through the integrated service control point.

Detailed Description Text (7):
In a further aspect of the present invention, a method is provided for accessing service data relating to a subscriber's telecommunications services using a graphical user interface (GUI) through multiple data networks, and using an interactive voice response (IVR) system through a public switched telecommunications network. The method includes providing the subscriber with the option of accessing the service data through more than one interface, including the IVR system and the GUI, and the subscriber selecting either the IVR system or the GUI. The service data is accessed through an intelligent peripheral, which obtains the service data from an integrated service control point. The service data is presented to the subscriber through the selected interface, so that the subscriber can access the service data through the IVR system or the GUI, based upon the subscriber's selection.

Detailed Description Text (8):
According to another aspect of the present invention, a system is provided for reviewing and updating a subscriber's telecommunications services using a graphical user interface through multiple data networks. The system includes a Web client, through which the subscriber views service data received through the data networks and requests service data updates. The service data is viewed through a graphical user interface. The system further includes a Web server that receives a data message, which indicates a subscriber's desired update to a selected telecommunications service, transmitted from the subscriber in response to a service data update and an intelligent peripheral, which receives the data message via at least one of the data networks. The intelligent peripheral translates the data message into a standard protocol, which includes the SR-3511 protocol. The translated data message is identical to a data message that the intelligent peripheral would create if the subscriber had entered the desired update through an interactive voice response system. The system also includes an integrated service control point that receives the message in the standard protocol and updates the selected telecommunications service in accordance with the subscriber's desired update. The selected telecommunications service is updated in the integrated service control point substantially contemporaneously with the subscriber requesting the update at the graphical user interface. Furthermore, the subscriber retains the ability to update and review the service data through an interactive voice response. The system may include a service status database from which the service data is initially retrieved, thereby reducing traffic on the integrated service control point.

Detailed Description Text (9):
In another aspect of the present invention, a method is provided for accessing caller ID data relating to a subscriber's remote access to caller ID service using a graphical user interface (GUI). The method includes identifying selected telecommunications services managed by a personal call manager account belonging to the subscriber, at least one which is the remote access to caller ID service. The telecommunications services are presented to the subscriber at the GUI through at least one data network. The subscriber then queries an intelligent peripheral through the data network indicating the subscriber's desire to access the remote access to caller ID service. The caller data is then retrieved from a call logger database, which stores the caller ID data, in response to the query. The caller ID data is transmitted to the subscriber through the data network and is displayed at the GUI.

Detailed Description Text (10):
In a still further aspect of the present invention, a method is provided for providing caller ID information associated with a telephone call from a calling party to a destination, the caller ID information being provided over multiple networks to a subscriber at a location remote from the destination. The method includes storing caller ID data in a call logger database in response to the calling party placing the telephone call to the destination. A caller ID query is received from the remotely located subscriber through at least one of the networks. In response to the caller ID query, the caller ID data is retrieved from the call logger database, transmitted to the remotely located subscriber through at least two of the networks and displayed at the remote subscriber's location.

Detailed Description Text (11):
The method for providing caller ID information to a subscriber at a location remote from the telephone call destination may further include initially launching an AIN trigger when the calling party places the telephone call to the destination which subscribes to a remote caller ID service. In that case, the storing of caller ID data includes transmitting calling party

information associated with the calling party from an integrated service control point to a GDI server, obtaining additional information from a directory server based upon the calling party information and transmitting the caller ID information from the GDI server to the call logger database. The additional information can be obtained from the directory server by either the GDI server or by the integrated service control point, which forwards the additional information to the GDI server. The caller ID information may include the calling party information and the additional information.

Detailed Description Text (12):
The method for providing caller ID information to a subscriber at a location remote from the telephone call destination may also include determining whether the subscriber has activated the remote caller ID service. Also, at least one of the networks may be a packet switched data network, which may include the Internet. Also, receiving the caller ID query may include receiving at a Web server the caller ID query from the subscriber through a Web client, so that transmitting the caller ID data to the remotely connected subscriber includes transmitting the caller ID data from the Web server to the web client.

Detailed Description Text (13):
In another aspect of the present invention, a system is provided for providing caller ID information, associated with a telephone call from a calling party to a destination, to a subscriber at a location remote from the destination. The system includes an advanced intelligent network (AIN), which includes an integrated service control point that forwards calling party information in response to the telephone call, and a private network, which includes multiple servers in communication with one another. A first group of servers forwards caller ID information based upon the received calling party information, to a call logger database. The system further includes a public network, including a client which sends a caller ID query to a second group of servers. The public network retrieves the caller ID information from the call logger database and sends the caller ID information to the client. The subscriber can view the caller ID information while being located remotely from the destination of the telephone call associated with the caller ID information. The public network may be the Internet and the client may be a Web browser.

Detailed Description Text (14):
According to another aspect of the present invention, a system is provided for providing caller ID information, associated with a telephone call from a calling party to a destination, to a subscriber at a location remote from the destination. The system includes a switch, associated with the destination, that receives the telephone call from the calling party. The switch has an AIN trigger set to launch a query in response to the telephone call. The system further includes an integrated service control point that forwards calling party information in response to the query and an interface server that obtains additional information from a directory server, based upon the received calling party information. The caller ID information includes the additional information and the calling party information. The system further includes a call logger database that receives the caller ID information from the interface server and stores the caller ID information. The system also includes a Web client that forwards a caller ID query from the subscriber and a Web server that receives the caller ID query from the Web client over the Internet and, in response to the query, retrieves the caller ID data from the call logger database and forwards the caller ID data to the Web client for display to the subscriber. The subscriber can view the caller ID information while being located remotely from the destination of the telephone call associated with the caller ID information.

Detailed Description Text (15):
The present invention is an AIN based system and method that allows a PCM subscriber connected to a communications network, including the Internet and other packet switched type data networks, as well as through conventional IVR systems, to customize and execute services associated with telephonic communications, with near real-time access to the service data. FIG. 1 illustrates an exemplary telecommunications network (e.g., PSTN) in association with the present invention. The network includes a calling party 20, an originating Service Switching Point (SSP) 21, a terminating SSP 24 and a subscriber's telephone (i.e., the destination) 25. The network also includes a Signaling Transfer Point (STP) 22, an Integrated System Control Point (ISCP) 23 and an Advanced Intelligence Network-Intelligent Peripheral (AIN-IP or intelligent peripheral) 40. The intelligent peripheral 40 includes an interactive voice response (IVR) system. By way of example, the ISCP 23 may be implemented with the Bellcore

Integrated Service Control Point, loaded with ISCP software Version 4.4 (or higher), available from Telecordia, Murray Hill, N.J.

Detailed Description Text (17):
The user is able to access the intelligent peripheral 40 through the Web server 43, which is in communication with the Internet 44 or other packet switched data network. The user is alternatively able to access the intelligent peripheral 40 through the IVR system 45 using a conventional DTMF telephone connection. When using the Internet, the user accesses the Web server 43 with a PC, acting as a Web client 30, using software such as ICW Client, available from Southwestern Bell Telephone Company. The Web client may likewise incorporate a Web browser, such as Microsoft Internet Explorer, available from Microsoft Corporation, or Netscape Navigator. In one embodiment, the Web client 30 is implemented with an IBM Pentium based computer, running the Linux or Microsoft Windows operating system and the Microsoft Internet Explorer, Netscape Navigator or Hotjava, available from Sun Microsystems, Inc., Web browser software. An embodiment of the invention with respect to the Web server 43 may include running the Linux or Microsoft Windows operating system and the Apache Web server software, available from the Apache Software Foundation, or the Jigsaw Web server software, available from World Wide Web Consortium (W3C).

Detailed Description Text (18):
The SSP 24 is the terminating central office (CO) for the PCM subscriber 25 and the SSP 21 is the originating CO for the calling party 20. However, the terminating CO and the originating CO may be the same. The SSPs 21 and 24 may comprise, for example, 1AESS or 5ESS switches manufactured by Lucent Technologies, Inc., or DMS-100 switches manufactured by Nortel Networks Corporation (Nortel), or AXE-10 switches manufactured by Telefonaktiebolaget LM Ericsson.

Detailed Description Text (20):
FIG. 2 is an exemplary call flow diagram depicting a subscriber using the PCM service. Initially, the subscriber accesses a public packet switched data network, such as the Internet, from a Web client 30, using a Web browser such as Microsoft Internet Explorer, Netscape Navigator or HotJava. Once on the Internet, the subscriber connects to the Web server 43, which serves as a secure access platform. The Web server 43 receives HyperText Transfer Language Protocol (HTTP) messages from the Web client 30 and provides HyperText Mark-Up Language (HTML) Web pages in response to the subscriber's input to the Web client 30. The Web pages relate to the subscriber's PCM account.

Detailed Description Text (21):
Once connected to the Web server 43, the user must first log-in to the PCM account, also depicted at block 301 in FIG. 3 and described below. The log-in equates to an authentication of the user. To perform the authentication, the Web server 43 contacts the Authentication/Subscription Information (ASI) Server 42, which confirms that the subscriber is an authorized user by verifying at least the subscriber's name and a password. The ASI Server 42 also provides to the Web server 43 a list of the services to which the user has subscribed in the PCM account. Services for each phone number are linked to the PCM account through the ASI Server 42.

Detailed Description Text (23):
In another embodiment, depicted in FIG. 2A, the Web server 43 retrieves from a Service Status Database 41 the data and status of the various services managed through the PCM account, rather than from the ISCP 23, directly. This database serves as a cache for the service information in the ISCP 23. The Service Status Database 41 contains information current to the most recent update interface with the ISCP 23. The cache arrangement enables the user to efficiently access this information without waiting for the ISCP 23 to process the request. At the same time, it reduces ISCP traffic. The Service Status Database 41 is refreshed periodically to ensure currency, as well as pursuant to specific command by the user. This database is a conventional Lightweight Directory Access Protocol (LDAP), such as the LDAP available from Lucent Technologies, Inc. In the alternative, the database may be a standard relational database, such as those available from Oracle Corporation or Sybase, Inc.

Detailed Description Text (25):
At this point, the user may choose to update or to simply review the service information. When the service is updated, the Web server 43 sends the update instructions in a data message to the intelligent peripheral 40. The intelligent peripheral then translates the update

instruction into the SR-3511 protocol and communicates the updated service parameters directly to the ISCP 23.

Detailed Description Text (26):
For example, one available service is Incoming Call Manager (ICM), by which the user may prioritize, forward, preview or block selected telephone numbers. In the update procedure, the user enters a telephone number to be blocked, for instance, which the Web server 43 communicates to the intelligent peripheral 40. The intelligent peripheral, in turn, sends the data via SR-3511 to the ISCP 23, which flags the number to be blocked. Because the intelligent peripheral's instructions to the ISCP 23 are sent and implemented immediately, without the involvement of the provider's account management or customer service, the changes to the service are operable and available shortly after the user sends the instructions. In an embodiment that includes the Service Status Database 41, the cache will then be updated in due course to reflect the updated information in the ISCP 23.

Detailed Description Text (27):
FIG. 3 depicts the procedure followed by the subscriber when first entering the PCM Web site. The subscriber must first log-in at block 301. Assuming the subscriber's PCM account has already been established, as described below, he or she must provide the authentication data to proceed. The authentication data is entered at a log-in screen, an example of which is depicted in FIG. 4 at screen 401. To maintain the integrity of the secure platform, authentication requires preferably a user ID and a password. The user ID is any name, not necessarily unique within the PCM system, selected at account initiation by the subscriber. The password is confidential (at the subscriber's discretion) and must be unique with respect to the associated user ID. The subscriber may change the password as desired, but appropriate authentication data must be provided prior to such changes. If the subscriber enters an invalid user ID or password, the Web server 43 responds with a message explaining the problem and allows another chance to enter correct data.

Detailed Description Text (28):
After the subscriber is authenticated, the subscriber proceeds to enter the PCM at block 302. At this time, the user views a general informational screen 402, which is formatted at the discrettion of the service provider. It may include, by way of example, new services offered to the subscriber. After the subscriber elects to proceed into the PCM, the Web server 43 navigates to a page 404 that displays telephone numbers associated with the PCM account(s) to which the user belongs and to which the user is authorized to access. FIG. 5 depicts an exemplary screen displaying phone numbers to which the user has access. At this point, the user selects a telephone number at block 303 and the corresponding services are displayed for the selected telephone number at screen 403. The user may then elect to implement the various services in place for a particular phone number or, depending on the user's privileges within a particular account, such as a superuser or a PCM user (described later), to manage the PCM account.

Detailed Description Text (29):
If the PCM has more than one associated telephone number, the user would see a Web page listing the numbers, as in block 404 of FIG. 4. The screen has user interface elements that allow the user to select one of the numbers. Thus, each PCM account keeps track of a nonempty set of phone numbers to be managed through the PCM on behalf of the corresponding set of users, presumably members of a family, business, organization or other group.

Detailed Description Text (30):
After the user selects a phone number at block 303, the system displays for the user a PCM summary page 304 corresponding to the selected telephone number. The PCM summary page displays only data the user is authorized to see for the selected telephone number. As shown at screen 403, the PCM summary provides various options to the user, including by way of example, selecting from among listed services 306-309, returning to select an alternative PCM telephone number or exiting PCM altogether 313.

Detailed Description Text (31):
FIG. 6 shows an exemplary PCM summary display, which corresponds to screen 403 of FIG. 4, entitled Personal Call Manager Home Page for account number (512) 555-5831, which is the selected telephone number in the example. FIG. 6 shows four services accessible through the PCM, although the four services are not intended to be limiting. That is, the PCM is able to

administer any call services associated with an ISCP 23. The services depicted in the PCM summary screen 403, as well as in FIG. 6, are Caller ID Log 601, Message Center 602, Incoming Call Manager (ICM) 603 and Outgoing Call Control (OCC) 604. The displayed information is summary in fashion, the details being available to the user through selection of one of the available services, which displays a PCM Service screen 405. In the depicted embodiment, the summary as well as the detailed data is provided through the Web server 43 from the ISCP 23 or a Call Logger Database 95 (shown in FIG. 8 and discussed in detail below), depending on the service selected. At the summary screen, the Caller ID Log 601 retrieves data from the Call Logger Database 95 and shows, for example, the number of call records added since the last review and the Message Center 602 likewise retrieves data from the Call Logger Database 95 and shows the number of new call notes, e-mails, wireless calls, faxes and messages reviewed. The ICM 603 retrieves data from the ISCP 23 and shows the status of the call blocker, call forwarding, priority call and caller preview features and an Outgoing (OCC) summary retrieves data from the ISCP 23 and shows whether international calls, long distance calls, 900/976 numbers and/or directory assistance calls are restricted.

Detailed Description Text (33):
The paging capability provides the option of paging the subscriber when a Caller ID is received from a subscriber specified phone number. Paging may include a page, a wireless short message, an email, or a generated phone call to a specified number. Moreover, Caller ID logs can be collected and paged to the subscriber at periodic intervals with summary and/or detailed information.

Detailed Description Text (35):
In one embodiment of the invention, a Remote Access Caller Identification (RACLID) service is offered as one of the services available to the user. The RACLID service permits subscribers to access their Caller ID data when they do not have access to customer premises equipment, such as their Caller ID box. Conventional implementation of Caller ID presupposes delivery over the subscriber's telephone line to a Caller ID box attached to that line. According to the invention, the Caller ID data is delivered via the data networks (including, for example, the Internet) to the subscriber. Where RACLID is available and incorporated into the PCM, it is specifically listed as one of the selectable services at the PCM Summary page 403.

Detailed Description Text (36):
As in the case of the other services in PCM, the subscriber can review caller data using RACLID from any location with networking facilities that allows connection to the data network on which the Caller ID data is stored. The networking facilities would include the Internet, a corporate intranet or other TCP/IP network. Also, RACLID may be provided without PCM.

Detailed Description Text (38):
Outside the PSTN network, the RACLID service requires multiple servers and databases, also depicted in FIG. 9. These elements include a GDI server 92, a directory server 93, a Web server 43 and a Call Logger Database 95. Generally speaking, the GDI server 92 interfaces with the GDI client 91, facilitating communication between the PSTN and the RACLID networks. The directory server 93 contains information stored by the RACLID provider, including data associated with the accessible universe of telephone numbers, regardless of whether they are associated with RACLID subscribers. It also stores authentication data corresponding to RACLID subscribers. In an embodiment of the invention, the directory server 93 can be incorporated into the AIS server 42, which contains authentication data corresponding to PCM in general. In another embodiment, the information stored at the directory server 93 may be incorporated into the Web server 43. The Web server 43 connects the subscriber through the Web client 30 via a data network 44, such as the Internet, and conducts the various interactive operations with RACLID. In alternative embodiments, the Web client 30 is implemented with an IBM Pentium based computer running the Linux or Microsoft Windows operating system and a Web browser, such as Microsoft Internet Explorer or Netscape Navigator. The Call Logger Database 95 contains data associated with specific RACLID accounts. In an embodiment of the invention, the Call Logger Database can be incorporated into the Service Status Database 41.

Detailed Description Text (39):
Referring to FIG. 9, the RACLID system is initiated by an AIN Terminating Attempt Trigger (TAT) launched by the SSP 24 whenever a call is placed by the call originator 20 to a RACLID subscriber's phone 25. Once the trigger has been assigned and activated, every call terminating to the PCM subscriber's line will cause the SSP 24 to launch a TAT query via the existing

Signaling System 7 (SS7) network (and appropriate STP 22) to the ISCP 23. The SSP 24 is the terminating central office for the RACLID subscriber. The TAT is assigned to the RACLID subscriber's directory number or line, depending upon the type of switch. Significantly, the call is not suspended at the switch during execution of the RACLID process. Rather, the call completes in a normal fashion.

Detailed Description Text (42):
When the GDI server 92 receives the InvokeApp message from the ISCP 23, it first determines whether the subscriber has activated the RACLID service. This avoids unnecessary processing time in the event the RACLID service is OFF. The GDI server 92 accomplishes this by querying the Call Logger Database 95, which contains an ON/OFF indicator dedicated to the subscriber's account. To activate the system, the subscriber accesses the Web server 43 by means of the Web client 30. The Web server 43 in turn accesses the Call Logger Database 95. The subscriber selects the ON option to activate the RACLID service, which then remains active until the subscriber accesses the Call Logger Database 95 and selects the OFF option. The subscriber may perform the ON/OFF commands through any means of access to the Web, as opposed to being limited to the phone number associated with the RACLID account. Note also that any conventional Caller ID system using customer premises equipment, such as a Caller ID box, by the same subscriber is unaffected by the ON/OFF command directed to the RACLID service.

Detailed Description Text (43):
The servers and database communicate with one another using Java Database Conductivity (JDBC), although any appropriate interface may be used. Also, alternative embodiments of the invention combine the various server and database functions into any combination of systems, including a single server. With respect to subscriber access to the system, the communication between the Web server 43 and the Web client 30 uses HTTP, although any appropriate interface may be used.

Detailed Description Text (44):
Once the GDI server 92 detects the active or ON status, it proceeds to contact the directory database server 93 to retrieve the calling party's name associated with the telephone number provided by the ISCP 23. In an embodiment, the directory database server 93 is a Line Information Database (LIDB) server. The LIDB server is maintained independently of the PSTN and updated appropriately by the service provider to assure provision of current information. The invention may include any comparable server, however, including a Lucent LDAP server. After the calling party name is retrieved from the LIDB, the GDI server 92 then provides the calling party's name, along with the caller data provided by the ISCP 23, including the calling party's number, the called number, the call date and the call time (collectively referred to as Caller ID information), to the Call Logger Database 95, where it is stored for later retrieval.

Detailed Description Text (45):
In an alternative embodiment (not pictured), the ISCP 23 contacts the directory database server 93 directly to retrieve the calling party's name associated with the calling party's telephone number. The ISCP 23 then sends the calling party's name to the GDI server 92, along with the calling party's number, the called number and the current date and time in the InvokeApp message. The GDI server 92 then provides this Caller ID information to the Call Logger Database 95, where it is stored for later retrieval. Although obtaining the information from the directory database server 93 somewhat more efficiently, this embodiment requires additional work by the ISCP 23.

Detailed Description Text (46):
In order to retrieve the Caller ID information from the Call Logger Database 95, the subscriber simply accesses the Web server 43 once again, through any means of access to the Web, and enters authentication data at the log-in. Assuming the subscriber is running the PCM, he or she then selects the RACLID option from the PCM Summary Web page 801 of FIG. 10, which is a block diagram of the various functions available to a typical user at the RACLID service Web page, discussed further below.

Detailed Description Text (47):
Alternative embodiments of the invention do not require specific incorporation of the PCM. For example, one embodiment enables RACLID subscribers, including those who do not necessarily have PCM, to go directly to a RACLID dedicated home page, which would be substantially similar in appearance to that depicted in FIG. 7.

Detailed Description Text (48):
Next, the Web server 43 receives the log-in information from the Web client 30 and queries the directory server 93 to retrieve authentication data corresponding to the user account. The authentication data includes a user identification and a password. The user identification is any name, not necessarily unique within the RACLID system, determined by the subscriber. In one embodiment, the user identification and password correspond to the user identification and password of the PCM. The Web server 43 then requests input of the authentication data from the subscriber and compares the input data with the directory server data to determine a match.

Detailed Description Text (49):
After authentication, the Web server 43 processes the commands entered by the subscriber. Available commands are depicted in FIG. 10 in the Call Log 802. One command sent automatically upon logging onto the RACLID service is the request data command, pursuant to which the Web server 43 retrieves all stored caller information from the Call Logger Database 95, including the calling party's name and corresponding phone number and the date and time the call was placed. This information is displayed as indicated, for example, in FIG. 7.

Detailed Description Text (50):
Another command is the delete data command 810, pursuant to which the Web server 43 removes the stored Caller ID information associated with a selected calling party from the Call Logger Database 95. If no delete command is executed by the subscriber, the Caller ID information stays in the Call Logger Database 95 and will continue to be retrieved pursuant to further request data commands until a delete command is sent or until some predetermined drop time expires, for example, 30 days. In one embodiment, the drop time may be adjusted by the subscriber.

Detailed Description Text (51):
Other interactive commands, shown in FIG. 10, include the delete all command 806 and the refresh display command 805 for the subscriber's convenience. Pursuant to the delete all command, the Web server 43 erases from the Call Logger Database 95 the Caller ID information currently and displayed. Pursuant to the refresh display command, the Web server 43 queries the Call Logger Database 95 with an updated request data command, which would retrieve any caller information received after execution of the previous request data command. The ON/OFF switch 803 is also provided on the display Web page to activate/deactivate the RACLID service.

Detailed Description Text (52):
In the PCM environment, RACLID data may be used to accommodate other optional services. For example, the Directory Entry service 807 of FIG. 10, when selected by the user, automatically deposits name and telephone number information in the user's personal directory. In an alternate embodiment, the data may populate data in a PDA, e.g., a 3Com Palm Pilot. The Voice Over Internet 808 automatically places a call to the selected telephone number over the existing network connection. The user can also select the return to PCM Summary option 811 to access other services.

Detailed Description Text (54):
In an embodiment of the invention, a user's interaction with the PCM, as well as the functionality of the PCM, is implemented with object-oriented programming, the terminology of which incorporates "classes" and "types." A class is a programming construct for defining the implementation of objects (e.g. users, telephone numbers, and services) that have the same sort of data and procedures. A type specifies the properties of a set of objects without reference to implementation. A type therefore can be implemented as many different classes. A type can also be implemented as a C++ class or as a JAVA class, and it can even be implemented in multiple ways within the same language.

Detailed Description Text (59):
Administrative interaction with the PCM account uses PcmAccount 1202 operations, which include for example setName, setID, setSuperUser, addUser, removeUser, addPhoneNumber, removePhoneNumber, addService and removeService, which enable the account to be defined. Management of a PCM account includes creating the account. To do so, the subscriber must provide necessary personal information to the PCM service provider in exchange for the unique log-in ID and name. The webmaster initiates the account using the setName and setID operations, respectively. The ID is essentially the password and therefore must differ from every other PCM account ID known to the system. The name, however, need not be necessarily unique.

Detailed Description Text (60):
In its initial state, a PCM account is additionally assigned a superuser (usually the subscriber), a set of authorized users (initially consisting only of the account superuser), a personal directory for the superuser and a set of one or more associated phone numbers. The webmaster sets the initial PCM account superuser using the setSuperUser operation. The superuser must be a member of the account and, as stated above, has read and write privileges to the account. Either the webmaster or the account superuser can then add new users to the account if a new user is not already a member using the addUser operation. After an operation adding a user is complete, the PCM account and the new user will cross reference each other in the attributes of User 1102 and PcmAccount 1202, which information is stored in the ASI server 42 in one embodiment.

Detailed Description Text (61):
This operation does not provide the new user access to any of the account telephone numbers, which is performed separately under an AddPhoneNumber operation on a phone number by phone number basis. This data is likewise stored at the ASI server 42. Both the webmaster and the superuser provide access to the specified phone number within the PCM account. The telephone number and user must already belong to the account.

Detailed Description Text (62):
The webmaster or the superuser can likewise remove a user from an account if the user is already a member of the account using the removeUser operation. The user to be removed cannot be the account's superuser. After the operation completes, the cross-references in the attributes of User 1102 and PcmAccount 1202 are eliminated and all telephone number access privileges are revoked.

Detailed Description Text (63):
The webmaster can add a telephone number to a PCM account if that phone number is not already in the account. Again, after the operation completes, the account and the new phone number will have references to each other in the attributes of User 1102 and PcmAccount 1202 stored at the AIS Server 42. The added phone number will necessarily include the account's superuser in its list of allowed users. The webmaster can likewise remove a phone number from a PCM account if that phone number already belongs to the account using the removePhoneNumber operation. After the operation completes, neither the account nor the telephone number have references to each other in attributes of User 1102 and PcmAccount 1202.

Detailed Description Text (64):
The webmaster can perform other functions in the PCM account, as well. For example, the webmaster can add a service to a telephone number in an account if that service is not already present on that phone number. The webmaster can likewise remove a service from a phone number in an account. Also, the webmaster can add or remove a personal directory to the account.

Detailed Description Text (73):
Although the present specification describes components and functions implemented in the embodiments with reference to particular standards and protocols, the invention is not limited to such standards and protocols. Each of the standards for Internet and other packet switched network transmission (e.g., TCP/IP, UDP/IP, HTML, SHTML, DHTML, XML, PPP, FTP, SMTP, MIME); peripheral control (IrDA; RS232C; USB; ISA; ExCA; PCMCIA), and public telephone networks (ISDN, ATM, xDSL) represent examples of the state of the art. Such standards are periodically superseded by faster or more efficient equivalents having essentially the same functions. Accordingly, replacement standards and protocols having the same functions are considered equivalents.

CLAIMS:

2. The method of claim 1, further comprising determining whether the subscriber has activated the remote caller ID service.

5. The method of claim 4, wherein receiving the caller ID query further comprises receiving, at a Web server, the caller ID query from the subscriber via a Web client; and wherein transmitting the caller ID data to the remotely connected subscriber further comprises transmitting the caller ID data from the Web server to the web client.

6. A method for implementing a remote access to caller ID service for a subscriber, the service providing caller ID information associated with a telephone call from a calling party to a destination of a subscriber, the caller ID information being provided over a plurality of networks to the subscriber at a location remote from the destination, the method comprising: obtaining, at a service control point, calling party information associated with the calling party from a switch, the calling party information comprising at least a telephone number associated with the calling party; transmitting from the service control point to a GDI server the calling party information, while the service control point continues to process the call; obtaining, at the GDI server, additional information associated with the calling party from a directory server, the additional information comprising at least a name associated with the telephone number of the calling party; transmitting the caller ID data, comprising the calling party information and the additional information, from the GDI server to a call logger database; receiving a caller ID query from the remotely located subscriber via at least one of the networks; retrieving the caller ID data from the call logger database in response to the caller ID query; transmitting the caller ID data to the remotely located subscriber via at least two of the networks; and providing the caller ID information at the remote subscriber's location.

10. A system for implementing a remote access to caller ID service for a subscriber, the service providing caller ID information associated with a telephone call from a calling party to a destination of a subscriber, the caller ID information being provided over a plurality of networks to the subscriber at a location remote from the destination, the system comprising: a service control point that receives calling party data from a switch, associated with the subscriber destination, in response to the telephone call, the calling party data comprising at least a telephone number associated with the calling party; a GDI server that receives the calling party data from the service control point, while the service control point continues to process the call, the GDI server obtaining additional data associated with the calling party from a directory server, the additional data comprising at least a name associated with the telephone number of the calling party; a call logger database that receives the caller ID information from the GDI server, the caller ID information comprising the calling party data and the additional data; a network server configured to receive a first caller ID query from a client over a first network and, in response to the caller ID query, retrieves the caller ID information from the call logger database and forwards the caller ID data to the client; and an interactive voice response (IVR) configured to receive a second caller ID query from a telephone over a second network and, in response to the second caller ID query, retrieves the caller ID information from the call logger database and forwards the caller ID information to the telephone; wherein the subscriber can obtain the caller ID information from the server and the IVR while being located remotely from the destination of the telephone call associated with the caller ID data.

12. The system for implementing the remote access to caller ID service according to claim 11, wherein the second network comprises a public switched telephone network.

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